ACQUISITION, REDUCTION, AND ANALYSIS OF ACOUSTICAL DATA

AN UNCLASSIFIED SUMMARY OF
ACOUSTICAL WORKING GROUP STUDIES
NADC REPORT NO. AWG-SU

DEPARTMENT OF THE NAVY
NAVAL AIR DEVELOPMENT CENTER
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This book is dedicated to the first Director and the Technical Director of the Defense Communications Planning Group, later renamed the Defense Special Projects Group, LTGEN Alfred D. Starbird, USA(RET) and Mr. David Israel, and to their respective successors, GEN John D. Lavelle, USAF(RET) and LTGEN John R. Deane, Jr., USA, whose vision and encouragement charted the mission of the Acoustical Working Group.

DEPARTMENT OF THE NAVY
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ABSTRACT

With the ending of American military involvement in the Indochina conflict it became appropriate to summarize the unclassified aspects of those investigations which might have a reference value to future scientific workers both in the civilian and military endeavors. The investigations of the Acoustical Working Group (AWG), a committee of knowledgeable persons from various civilian and Department of Defense agencies responsible for performing acoustical studies relevant to the Indochina conflict, clearly fall in this category. They embrace, in general, the study of sound propagation, a subject important to noise abatement, hearing protection, and other ecological and environmental concerns. Some of these studies (more than 50 formal reports were prepared in a period of five years) also are significant in a general academic sense and as such have been selected to appear in this volume. The topics covered are data acquisition, reduction, analysis, acoustical propagation, and the characteristics of sound sources.

Benjamin B. Bauer, Technical Editor
Jack R. Harris, Chairman

FOREWORD

In early 1967 a committee of knowledgeable persons from industry and the Department of Defense was formed under the title of "Acoustical Working Group" (AWG), its chairman reporting to the Naval Air (later Electronics) Systems Command. The functions of the Group were to conduct studies and to advise on matters relating to acoustics, with special emphasis in the areas of sound propagation, reception, and signal processing and associated engineering disciplines. Formed to meet the needs of certain classified projects in the Department of Defense, the Group operated in a manner similar to the divisions of the National Defense Research Committee (NDRC) of World War II. The Group's original membership comprised:

Mr. Benjamin B. Bauer - CBS Laboratories
Mr. Edward T. Hooper - Naval Air Systems Command
Mr. Rowland H. McLaughlin - University of Michigan
Dr. John C. Munson - Naval Air Systems Command
Mr. Joseph Petes - Naval Ordnance Laboratory
Mr. Forrest C. Titcomb - Naval Research Laboratory
CDR Jack R. Harris - Naval Air Development Center, Chairman

Later, as program emphasis changed, the Group was expanded to include:

Mr. Edward J. Foster - CBS Laboratories
Dr. Paul E. Grant - Department of Defense
Mr. Donald Grogan - Picatinny Arsenal (Army)
Mr. Robert F. Hand - University of Michigan
Mr. C. William Hargens - Franklin Institute (Consultant)
Mr. James R. Howard - Naval Air Development Center
Mr. Sidney Krieg - Naval Air Development Center
LT Robert H. Marks - Department of Defense
Mr. Richard G. Satz - Picatinny Arsenal (Army)
Dr. Tsute Yang - Villanova University (Consultant)

Though the group as originally constituted was dissolved when its mission was completed, some of its studies were of sufficient general interest to warrant republication in a summary fashion. This report encapsulates, as appropriate, all of the unclassified aspects of AWG studies including procedures and results, with Chapters 1 and 2 previously separately reported. Classified summary results are expected to be reported at a later date. The technical editor appointed for these efforts was Mr. Benjamin B. Bauer of CBS Laboratories. He was responsible for editing the material contributed by appropriate AWG members.

JACK R. HARRIS
Scientific Officer
Naval Air Development Center

Acoustical Working Group Studies

ACQUISITION, REDUCTION, AND ANALYSIS OF ACOUSTICAL DATA

PREFACE

This volume is a collection of techniques related to the science of data acquisition, reduction and analysis developed and used by the Acoustical Working Group in the course of its studies, together with a summary of the essential theoretical principles needed to understand their proper use. A catalog of the data gathered by AWG in the course of its studies is provided. And, since verbal descriptions cannot replace the aural experience, a 7-inch, 33-1/3 rpm disk record containing some of the exotic, as well as common, sounds studied by AWG is included in a pocket inside the back cover of the book. Access to the cataloged data can be obtained by persons with justified requirements upon application to the Commander, U. S. Naval Air Development Center, Warminster, Pennsylvania.

The publication of this volume at this time is believed to be of special importance in view of the burgeoning interest in the practice of noise control and environmental acoustics, which is so heavily dependent on accurate measurements and meaningful data reduction. Admittedly, much of the specific information discussed here will have no direct application to home environments, nevertheless, the experience documented in this volume will prove to be extremely useful to the civilian practitioner. For one, under a single cover the text offers a complete description of the methodology of obtaining accurate and reliable acoustical data and manipulating it into useful formats. To an executive responsible for decision-making in connection with acoustical studies, such a compendium is of immense benefit, for

it helps him to assess realistically the magnitude of a given task and to chart accurately its successful execution. To the personnel planning the details of the project, or going into the field without extensive prior sound-measurement experience, this volume provides detailed guidance on how to select and use equipment, and how to maintain proper records and implement calibration procedures--which may well spell the difference between success and failure. And, to the data analyst, the text offers a variety of alternative methods for reduction and presentation of data, and suggestions for interacting with the data gathering teams for maximum information yield per unit effort.

Reviewing briefly the contents of this volume, Chapter One, which previously had been published as Report No. NADC-AWG-S1 on 31 March 1969, is a collection of techniques and precautious measures that have been found useful in gathering acoustical data, especially under inclement field conditions. It encourages the practitioner to prepare carefully for the mission, to make specific plans, to devise and acquire suitable instrumentation, and in other ways to perform preparatory tasks which ensure that the desired results are effectively obtained.

Chapter Two, previously issued as Report No. NADC-AWG-S2 on 1 October 1971, begins with the classification of the various aspects of data reduction and analyzes their interface with the data gathering function. This is followed by a review of the theoretical aspects of various methods of frequency- and time-domain analyses, a description of the instruments and recommended practices for their use, and suggestions about presentation of results. It should be noted at this writing that modern computer technology has now provided us with new and sophisticated tools--such as the Fast Fourier Transform and the Integrating Frequency Analyzer--which were just coming into their own during AWG studies; nevertheless, the basic considerations in Chapter Two are as applicable today as they were during the early stages of AWG work.

Chapter Three is a compendium of the various aspects of sound-wave propagation that should be clearly kept in mind by all persons engaged in acoustical data acquisition, reduction, and analysis. This chapter summarizes the basic ray and the normal-mode theories and reviews the effects of temperature, wind, channeling, scattering, etc. that will be especially useful to those engaging in the studies of distant propagation of sound in urban and suburban settings. Examples of propagation behavior in the air are given.

Chapter Four develops a logical procedure for cataloging the results of acoustical studies and provides a useful compilation of examples of various types of sound studied by AWG. It also contains a selective catalog of the detailed reports prepared by AWG. And, finally, the disk record furnished with the volume provides the naturalist and the urbanologist alike with a sampler of interesting acoustical signatures.

Having been privileged to participate in the work of AWG since its inception, the Technical Editor cannot help but observe how useful this volume might have been had it been available to the Committee at the outset, rather than toward the end of its mission. This would have saved countless annoyances and made our task much easier. But this is akin to wishing that in youth we had the experience gathered during a lifetime. In providing this summary of AWG studies, we trust that future workers in noise control and environmental acoustics will profit by availing themselves of the techniques here outlined, and thus be able to perform more effective service on behalf of a quieter world.

A few words of credit are in order. What has distinguished AWG from other Committees and Boards in which this Technical Editor has participated is the resolute and tireless dedication of its members, all of whom have made significant contributions to the work of the Committee and to the preparation of its reports; with special mention due its indefatigable Chairman who always provided effective and inspiring leadership and invariably managed to see each task through even when all else seemed to fail. In addition, he was a principal contributor to numerous AWG reports and to this final volume. An expression of gratitude also is accorded to Sherman Levin of CBS Laboratories who managed the editorial and production work for many AWG reports and for this final volume.

Benjamin B. Bauer
Stamford, Connecticut
May 1974

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A 7-inch, 33-1/3 rpm disk record containing representative sounds studied by AWG is included in a pocket inside the back cover.

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CHAPTER ONE

DATA ACQUISITION

CHAPTER ONE: DATA ACQUISITION

1.1 INTRODUCTION

The development of a system of acoustical detection for identification and reaction requires the initial acquisition of acoustical data upon which the system parameters will be based. Much of this data will be obtained outdoors in all kinds of weather, at various remote locations, and often under difficult and even hazardous conditions. These circumstances present the acoustical surveyor with a whole new set of problems normally not encountered in the laboratory. This chapter describes the principles of good engineering practice necessary to assure the acquisition of accurate and reliable data in remote areas under extreme environmental conditions. The information contained in this chapter is based to a large extent upon experience acquired during the performance of Acoustical Working Group (AWG) missions. The chapter can be used as the basis for practical data gathering missions with appropriate modifications to suit the particular circumstances.

1.2 PREPARING FOR AN ACOUSTICAL DATA GATHERING MISSION

A. Analysis of Objectives

The basic, but often neglected, task before starting a data acquisition mission is a careful analysis of objectives, which must be related to time and equipment limitations and to the problems that may be expected in carrying out the mission. Once a test is started it may become necessary to forego some objectives for others and, therefore, priorities should be attached to the various objectives. For example, a recording technique might be needed for sound detection and analysis which is different from that used for sound reproduction and simulation. If a choice must be made, it should be clearly understood which

objective of the mission is the more important to avoid possible disappointment.

An equally important aspect of the ultimate objectives is the early recognition of the type of data reduction that will be required. Unless the data reduction program has been planned and its needs established prior to collecting the data, it is very likely that more data than needed will be collected, but the information content of the data may be insufficient for the needs of the program.

B. Recognizing the Need for Special Equipment

Frequently, the type of terrain and the nature of the test make it necessary to provide special instrumentation which must be designed and fabricated before the data acquisition task begins. Thus, as much advance notice as possible should be given to allow for readying the equipment. In one instance, before acquiring the signatures of certain vehicles, it was ascertained (fortunately in time to take remedial action) that the radio transmission link introduced frequency response distortion which required the construction of compensatory devices prior to recording. This corrective action taken during the definition and planning phase enabled the mission to succeed where otherwise it might have failed.

C. Utilizing Practice Runs

A preliminary, or "dry," run under conditions simulating as close as practicable the actual test is almost invariably found to be helpful during the definition and planning stages. The dry run reveals the limitations of the equipment and personnel and allows remedial action which would be difficult, if not impossible, to take in the field. It also provides an opportunity to train new team members and to optimize the goals of the mission in line with the capabilities of the available equipment and time. A dry run is especially useful when a new parameter is introduced into the tests. For example, in a test which included the combined measurements of aerial and underwater sounds, the dry run revealed that some commercial hydrophones do not have suitable electrostatic shielding. This factor is of no con-

sequence for a sensor meant to operate at a depth of 100 feet or more, but when used in shallow water in combination with land-based instrumentation, it was found that severe electrostatic pickup was encountered requiring construction of special amplifiers and shielding devices. The mission would surely have been a failure if this "dry" run had not been conducted during the planning stages.

D. Advance Information Desirable

Sometimes security considerations severely limit the availability ahead of time of the detailed information needed to carry out an acoustical data-gathering mission. In this circumstance, planning cannot properly begin until the team arrives at the test site. However, as much information as security allows should be sought in advance to help in planning and in determining the equipment to be taken to the site.

Contrary to a popular misconception, the weight and bulk of equipment necessary for conducting acoustical surveys can be considerable. To adequately prepare for expected contingencies, in addition, can involve very sizable quantities of material. For one recent data acquisition test, to adequately equip a four-man field team it was necessary to commercially airlift 2800 pounds of equipment -- the largest single excess baggage load in the history of the airline.

1.3 PLANNING THE MISSION

A. General

The general plan begins with a study of objectives, consideration of the types of sounds to be recorded, selection and study of the site and terrain, decisions concerning equipment and personnel required, and the time period(s) during which survey information is to be obtained. The result of preparation of the plan is a Field Procedure (discussed in Section 1.5) detailing (1) how to conduct the particular test, (2) what equipment to provide, (3) the responsibility of various team members, and (4) what is expected to be contained in the data. The development of a Field Procedure requires considerable experience, and the following are some highlight factors which should be considered.

B. Site Selection

Ideally, the measurements should be made at a site and under conditions similar to those for which the end product is intended to be used. Frequently, however, this is not feasible and alternative sites must be selected. A site should be chosen which is operationally and acoustically similar to the ultimate use area. For example, the forest area should include similar trees to provide the same type of canopy, scrub, or open terrain as in the expected area of operation. Besides the forest area considerations, one should look for similar fauna and a similar background noise level as in the ultimate operation. Finally, security considerations and support considerations must be phased into the selection of a site. This is especially true in cases in which classified materiel is to be employed during the test.

C. Site Plan

During the preparation of the mission, a map of the site is most helpful. If one is not available, a sketch to approximate scale should be made at the earliest opportunity. In addition, photographs (especially aerial) are very helpful. Both the photographs and the map will be found useful not only for planning the test, but also for subsequent reduction of the collected data and, ultimately, preparation of the written report. The plan or map should indicate all the important details such as types of terrain and/or vegetation and any possible sources of anomalies and interfering sounds--hills and streams, buildings, etc.--which may shield the sound, create an echo, etc.

D. Dimensions, Distances, Parameters

Provision should be made for establishing dimensions and distances and for measuring other important variables. For example, if the purpose of the test is to measure the sounds of vehicles along a road, stakes or markers or other suitable sighting devices should be provided to enable the drivers and observers to read the positions as a function of time. If speed is one of the variables, then the plan should include means for determining and recording it. Things taken for granted often may foil a plan. For instance, all the preparations for a vehicle speed-

vs.-sound-level test may be defeated if the speedometer on the test vehicle is inoperative.

E. Data Sheet

A data sheet should be prepared which lists the acoustical events to be recorded, together with a checklist of important auxiliary information--wind, rain, time, etc. It should be remembered, however, that an observer busy operating a tape recorder and adjusting its gain for maximum information does not have the time to do much writing and a second man should be at the site to keep the written log. Since a written log may get separated from the tape, all comments should be recorded on a voice track which must be provided for this purpose. The major importance of the written log is as an aid in data reduction. A well documented log provides the analyst a means of rapidly locating key segments of data without the necessity of listening to the entire tape. For this purpose, correlation of the logs with a recorded time code is invaluable.

F. Climatic Conditions

Climatic conditions, the time of the year, and whether it is day or night have an important bearing upon the plan. Despite careful moisture-proofing, most equipment is subject to malfunction when left outdoors in the rain, or when high humidity combined with temperature changes causes moisture condensation. In this circumstance special precautions, such as frequent desiccation, are needed

G. Importance of Field Laboratory

Usually it becomes important to provide a field laboratory, suitably air conditioned, to permit the equipment to be maintained and to allow it to dry out and become stabilized and be recalibrated at intervals. The field laboratory also can be provided with equipment for preliminary scanning or taking a "quick look" at the data. A great advantage of such preliminary reduction is that the test plan may then be somewhat modified to insure that the ultimate objectives are met. A safe should be provided for storing all classified papers, tapes, and devices. Under unfavorable conditions when there is no time or opportunity to

provide a field laboratory, minor maintenance can be carried out in the field or in a hotel room. Under extremely unfavorable conditions--as with tests which extend for several uninterruped days in rain and humidity--less precise, but more sturdy, equipment may have to be used. The detailed requirements for the field laboratory are discussed later in Section 1.6.

H. The Personnel

The staffing of the team and assignment of tasks constitute an important part of the plan. Sending several people to a remote location involves considerable expense. On the other hand, it is usually not practical or advisable to field less than two people per site (or minimum of three if the equipment must be removed and redeployed each day). The age and health of the team members must be related to the stress involved, and such matters as security clearances, passports, insurance, vaccination, and even personal domestic problems may become formidable obstacles to an otherwise perfect plan. Experience in field procedures is, of course, extremely important and although not every member of the field team need have prior experience, at least two men per site should be fully familiar with the operation and limitations of the equipment.

I. Conclusion

The above brief description mentions only the most important factors to which the planner must address himself. The plan should always be evolved with the participation of the group leader who will be in charge of the team performing the tests and, if possible, with the advice of the more experienced field personnel. A more complete checklist for the items to be considered during a plan and the equipment needed for a mission are suggested later in this chapter, and in Appendix C.

1.4 EQUIPMENT CONSIDERATIONS

A. Custody of Equipment

One member of the team must be placed in charge of the equipment and should be provided with a list of items for which he will be accountable. In a multichannel system, the per-channel cost for

a suitable microphone, preamplifier, line driver, and auxiliary equipment runs in the thousands of dollars, and a multichannel portable tape recorder is also very expensive. Specially-built should be made to safeguard the equipment, to store it safely when the site is unattended, and to arrange for its deployment on site in a timely manner. Provision must be made for protecting both equipment and personnel from the weather. The shelter must be located sufficiently far from the sensor locations to prevent its affecting the data. For example, human activity will inhibit local fauna sounds and so the recording site should be at least 1000 feet from the microphones. During the AWG program, vans, tents, and station wagons have all been used as equipment shelters. The ideal situation is to have a van in which the equipment may be permanently mounted and moved from site to site.

B. Supplementary Tools and Equipment

Not only is it necessary to furnish safeguards for expensive equipment, but provisions should be made for each site to have a tool kit, flashlight and batteries, climbing equipment, and other items depending upon circumstances. One should consider those items which although within easy reach in the laboratory are inaccessible in the field.

C. Ranges of Level and Frequency

The expected ranges of level and frequency and type of data reduction desired play an important part in the selection of equipment for data acquisition. For example, FM recording preserves the low frequency response and amplitude and phase integrity of data, while direct recording provides greater high-frequency range capability and greater time-capacity-per-reel. Changing reels of tape during the data acquisition process always involves delays, and it is desirable to record only the minimum frequency range at the slowest possible speed that will realize the full potential of the test. Although this requires that the frequency range and S/N of the equipment exceed the range of the data, it is wasteful, for example, to record with a 10 kHz and 50 dB S/N capability if the sensor itself has a 1 kHz and 30 dB S/N range. On the other hand, it must be remembered that the maximum frequency ranges specified by the equipment manufacturers often strain the performance limits of the equipment, and therefore a generous margin for tolerance should be provided.

1.4.D-1.4.E

D. Measurement and Recording of Atmospheric Conditions

If the atmospheric conditions form an important part of the plan, provision should be made for their measurement and recording. As an illustration, if rain is part of the test, the measurement of rain in terms of inches per hour is a much more meaningful quantity than the mere statement "hard rain," "medium rain," etc. For this purpose, a simple rain gauge can be improvised of a tin can with its top removed. It is ideal if the meteorological data is recorded directly onto the magnetic tape.

E. Wind Effects

Wind is usually an important element, responsible for various types of noise. It creates turbulence around the sound measuring microphone. It induces movement of the microphone often resulting in that object coming in contact with others. It causes leaves to strike each other and excites resonances in open objects. By the interaction of cables and strings with the wind, the so-called "aeolian tones" are generated. All these effects should be carefully considered, together with an evaluation of their importance or decision about methods for coping with them in the field. For example, microphones and housings can be provided with windscreens, but whereas the windscreen affects microphone high-frequency response and, as has been discovered only recently, also affects the low frequency response of certain directional microphones, it must be preplanned and not makeshift.

Since wind is apt to be an important factor, usually it is also important to provide a means for measuring and recording its velocity vs. time. (As a minimum, provision should be made in the data log for making a notation of the approximate wind conditions, e.g. "slight wind causing the rustle of leaves.") Wind velocity can be measured with an anemometer. One type of gauge is dependent upon rotating vanes or buckets, but its inertia does not allow wind gusts to be measured accurately. An electrical gauge based on the hot wire principle permits the measurement and recording of the velocity of wind and wind gusts. Ideally, one data channel should be used to record the output of the hot wire anemometer when the objective of the test is concerned with wind activity, e.g. the determination of the effectiveness of a windscreen.

F. Multiplicity of Sites and Time Correlation

The question of using, say, four-channel recorders at two sites vs. one seven-channel recorder interconnected by long cables at one site depends upon equipment availability and the test parameters. In general, the attempt is made to minimize the number of individual recording sites, provided the test can be adequately monitored from a smaller number of sites.

When time correlation among various tape recorders is necessary, a master time code such as IRIG-B should be transmitted throughout the area and each recording site provided with a suitable receiver.

The distance between the microphone and the recorder is an important factor since, in connection with the expected signal levels and energy spectrum, it determines the power required to drive the cable. These factors should be considered in the plan and the necessary cables and line drivers fabricated and tested ahead of time inasmuch as a defective cable can cause major trouble in the field. More about this will be found in Section K, below, and subsequent sections.

G. Batteries, Illumination

Batteries for running the recording equipment present a special problem and an adequate supply of batteries and/or storage batteries and recharging facilities must be provided. If night operation is scheduled, suitable lights will be required.

H. Typical Arrangement of Apparatus

An illustration of the interconnections and apparatus for data gathering is shown in Figure 1-1. One to N microphones are placed in appropriate locations which must be an adequate distance away from the tape recorder to prevent site activity from affecting the data. Each microphone is connected to a transmission cable through an amplifier capable of driving the line. At the recorder, means must be provided for adjusting and monitoring the recording level. At least one track of the tape recorder is allocated for field operator commentary.

The prime calibration of the set-up is made using a known acoustic source at the microphones as seen in Figure 1-2. The source may be a General Radio (1562A) Calibrator, a pistonphone, or calibrated bullhorn loudspeaker to be described. By these means the signal level on the tape can be directly correlated

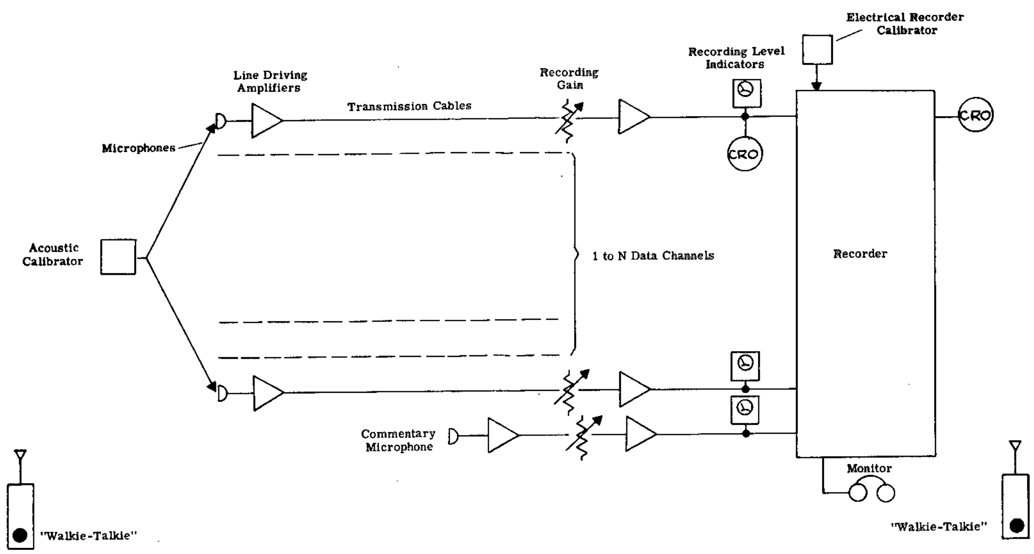


Figure 1-1. Photograph and Block Diagram of Typical Arrangement of Apparatus for Data Gathering

Figure 1-2. Calibration of the Equipment

with acoustic sound pressure level. A recorder calibrator capable of providing a known electrical signal to the recorder is frequently valuable both for recorder set-up and for establishing approximate acoustic level in those cases in which the field acoustic calibration is impracticable but where previous calibration data on the microphone is available.

Some means of monitoring the level of the recording is necessary to insure that the signal is well above the recorder noise level and below the overload point. Of course, the system noise level and overload point should have been previously established in the laboratory and documented for reference.

Earphones should be used in conjunction with the recording meter for monitoring the recording since one can usually determine the existence of an acoustic anomaly most readily by listening. An oscilloscope and/or peak reading storage meter such as an impact meter is ideal for monitoring peak factors which are not indicated on a meter. Such apparatus is imperative when recording impulsive data. It should be possible to monitor both before and after recording to determine that the data is, indeed, being properly recorded.

Finally, "walkie-talkie" communication between the field operator who may be calibrating the microphones and the tape recorder operator is a great asset.

I. Selection of Microphones

Microphone characteristics can be considered under two classifications. One of these concerns the directional characteristics of the microphone and the other concerns the frequency response and the particular type of transduction employed. In the former category, a distinction can be made between the omnidirectional (pressure) microphones, gradient (velocity) microphones, cardioid microphones, and other higher order directional arrays. For most acoustical data gathering conditions, the omnidirectional microphone is used. In some cases, however, the use of a directional microphone may be appropriate especially if one wishes to either maximize or minimize sound sources from a particular location.

The second method of classification is by transducer type, i.e. condenser, piezoelectric (ceramic) or dynamic. Each of the three types of microphones offers certain advantages as well as

disadvantages which must be considered. The condenser microphone (Figure 1-3) has the simplest physical construction and offers the best frequency response and phase linearity and lends itself readily to precise calibration. Because the diaphragm of the condenser microphone is extremely light and is resonated in the upper range of the bandwidth desired, the microphone is relatively insensitive to vibration. The main disadvantage of this type of microphone lies in its high source impedance which, unless special precautions are taken, makes it sensitive to moisture. This handicap limits the use of condenser microphones to laboratory application where the atmospheric conditions can be held under control.

The second type of microphone, the piezoelectric unit, is shown in Figure 1-4. Like the condenser microphone, it has flat frequency response at audio frequencies and therefore, in the field, it is sufficient to calibrate its response at a few points. Because of its lower source impedance, it offers improved moisture immunity over the condenser unit. The stiff and light transducer structure used insures low sensitivity to vibration. The piezoelectric microphone, such as the General Radio P-5 with suitable preamplifier and rain-and-wind screen (discussed in the following section), has been the principal data-gathering microphone in AWG work.

The final generic type microphone is the moving-coil dynamic outlined in Figure 1-5. This unit is rugged and moisture-resistant. Its source impedance is very low, allowing long cable runs without the need for preamplifiers. Because the frequency response is not as uniform as the previously described microphones, field calibration of the unit at a few points is insufficient and the use of an anechoic chamber is required. Furthermore, since the coil of the dynamic unit is relatively heavy and is resonated below 1000 Hz, it is more sensitive to mechanical vibration pickup than the other microphones. Nevertheless, under conditions of extreme humidity and continuous use without the opportunity of dessication and recalibration, the dynamic microphone (such as Model 655, especially moisture-proofed and anti-fungus-treated, manufactured by Electro Voice) has been found to be extremely useful.

J. Wind-and-Rain Screens

To minimize the effect of wind turbulence around the sensor, a special wind-and-rain screen for acoustical data-gathering microphones has been developed. As seen in Figure 1-6, a preamplifier is housed within the upper part of the enclosure and is protected by a water proof seal. Since the problem of wind noise can be extremely severe and flavor the acoustic data with the aero-

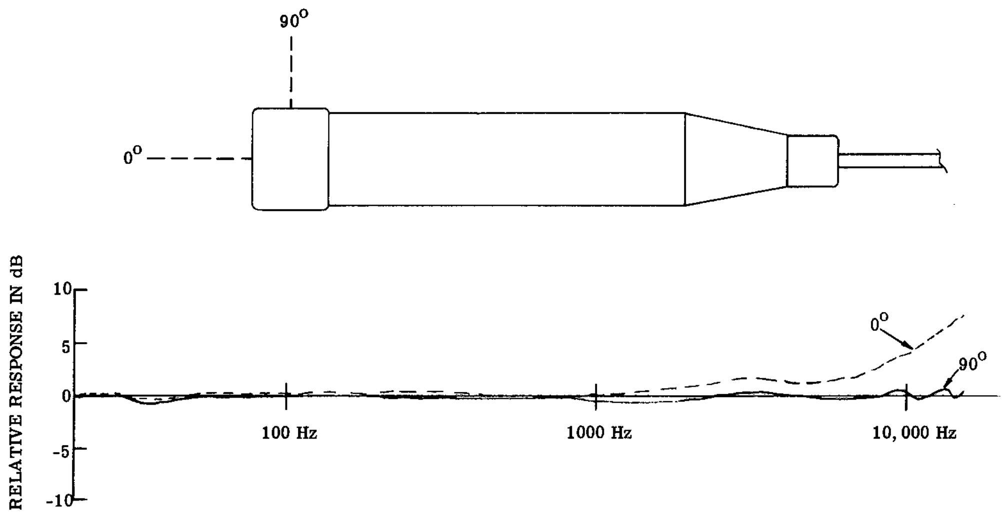


Figure 1-3. Typical Condenser Microphone (1/2" Diam.)

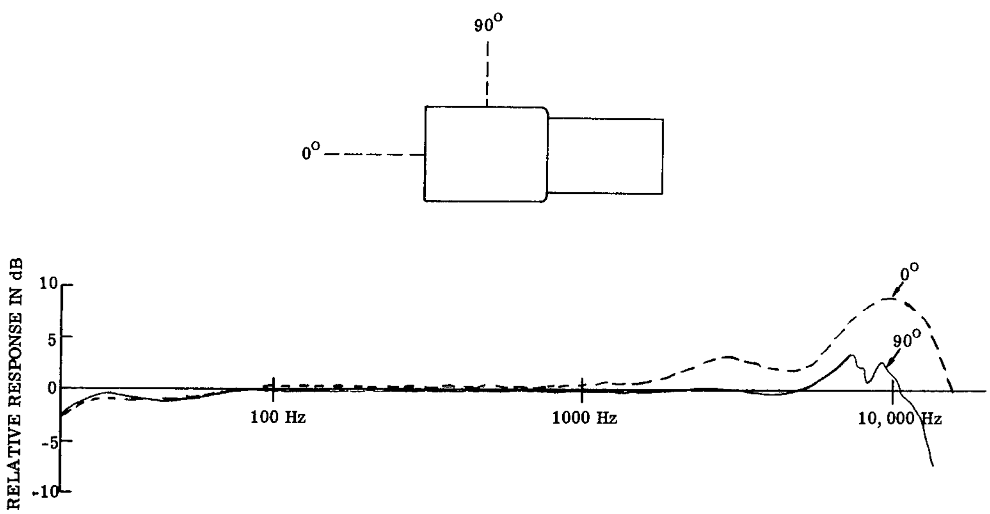


Figure 1-4. Typical Ceramic Microphone (15/16" Diam.)

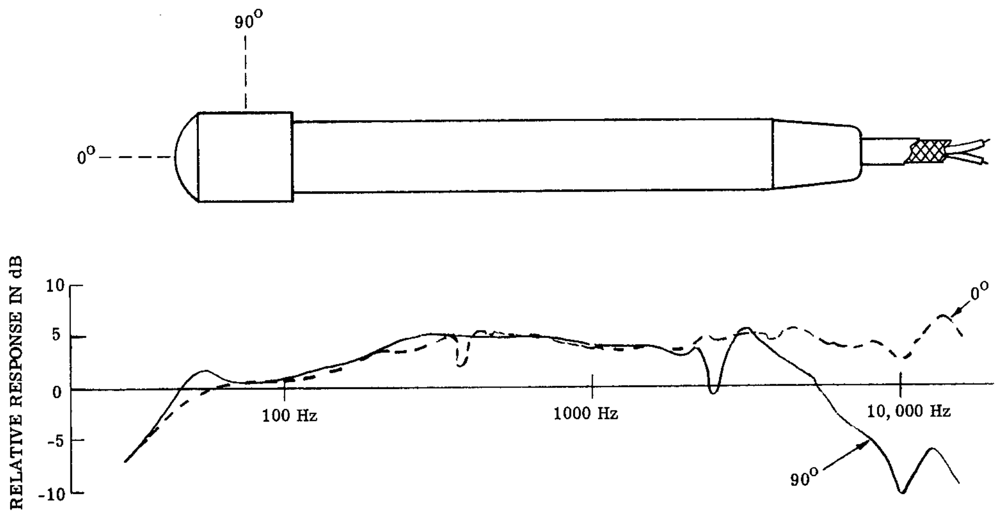


Figure 1-5. Typical Dynamic Mircophone (1-1/8" Diam.)

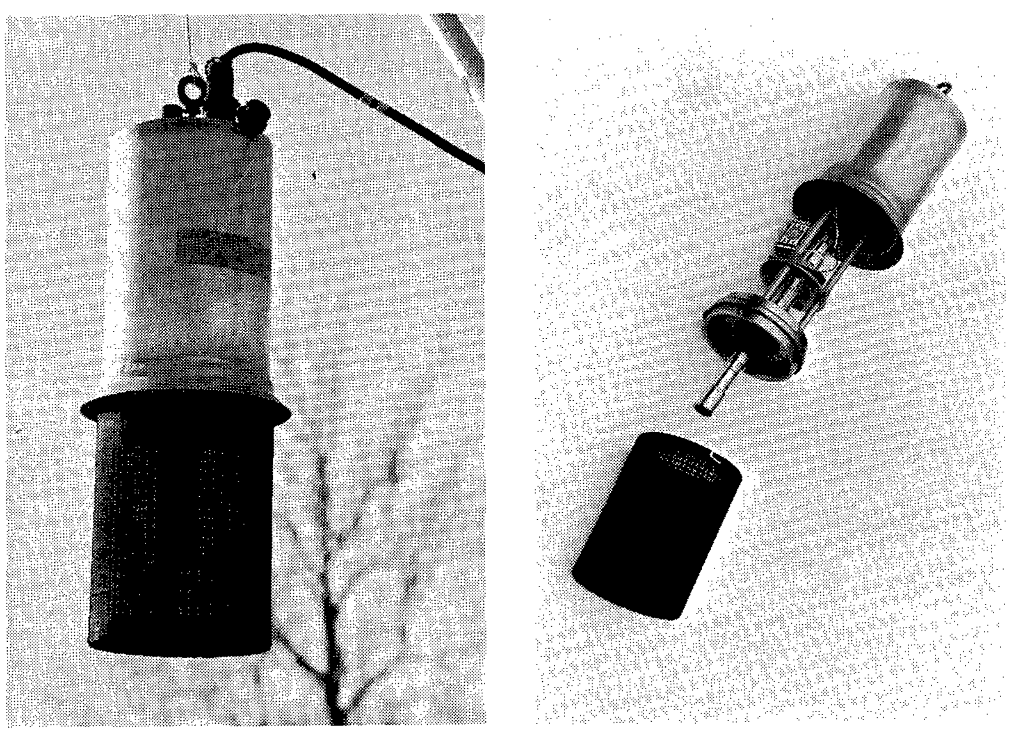


Figure 1-6. Wind-and-Rain Screen for the Acoustical Data-Gathering Microphone

dynamic shape of the sensor chosen, CBS Laboratories developed a windscreen testing machine capable of generating 30 mph wind. This apparatus (Figure 1-7) has been used in the development of a new acoustical data-gathering windscreen which offers considerable improvement over the unprotected microphone. As seen in Figure 1-8, in the critical lower frequency regions in which wind noise is most severe, a 25 dB improvement is noted. The windscreen portion is easily removed for acoustical calibration of the microphone (Figure 1-2). It is important to provide a microphone calibration in the anechoic chamber including the windscreen to determine how it affects the overall response.

K. Line Driving Amplifiers

In setting up the apparatus for acoustical data gathering the engineer will, of course, check the output impedance of his preamplifier to assure that the frequency response will not be degraded by the cable capacitance. With the use of liberal amounts of feedback in the preamplifier, the output impedance is usually very low and it would appear that long cables would not affect the accuracy of the data acquisition. What is frequently overlooked, however, is the capability of the amplifier to supply sufficient current required by the line capacitance. Very often the maximum output current capability of commercial preamplifiers is not specified.

Consider the problem of driving a 1000-foot cable with a 10 kHz signal whose magnitude is 1 volt rms. The cable capacitance is typically 50 pF/ft so that the total capacitance amounts to .05 μF. At 10 kHz, the reactance of this capacitor is about 310 ohms so that a peak drive current capability of 4.6 mA is required. Two questions, then, must be answered: Is the output impedance of the preamplifier much lower than 310 ohms so that the frequency response will not be degraded?; and, Can the amplifier supply 4.6 mA of current to drive the cable capacitance?. The answer to both of these questions must be "yes" if the system is to be used. Note that whereas the requirement on output impedance is independent of the signal levels, the drive capability must be considered a function of the expected signal levels. For example, if we want to use this amplifier to drive a 3-volt rms signal, it must be capable of producing peak currents of 14 mA.

In order to assure accurate data acquisition, a new complementary symmetry emitter follower line-driving amplifier has been developed (Figure 1-9). This amplifier accepts a single-ended input signal and uses a phase-splitter to drive identical output amplifiers, resulting in a balanced output configuration. A com-

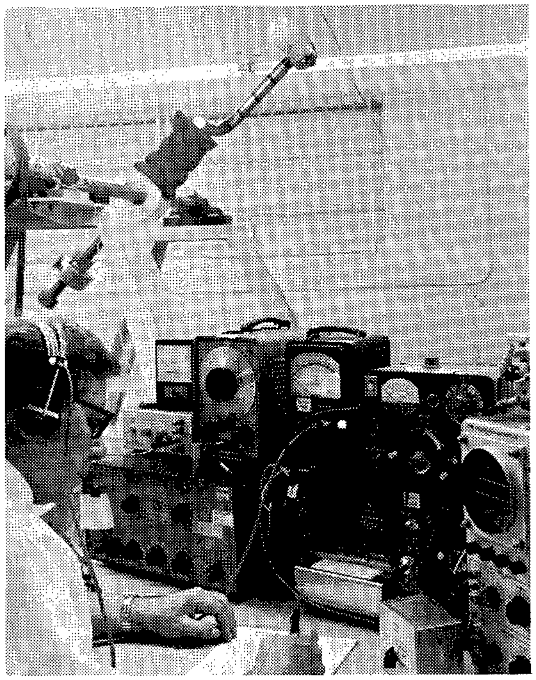


Figure 1-7. Windscreen-Testing Equipment

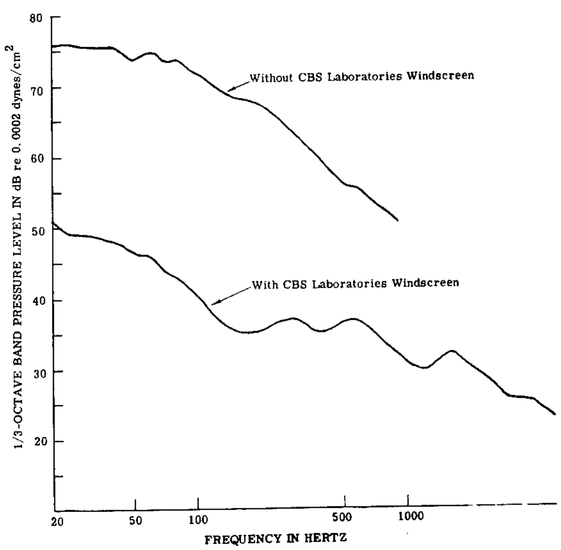


Figure 1-8. Windscreen Effectiveness (10 mph Wind)

plementary symmetry approach is employed to deliver the high peak currents required to charge the line capacitance. The outputs are, in effect, miniature class A-B amplifiers capable of delivering 100 mA peaks.

Figure 1-10 compares the capability of the new line-driving amplifier to that of a typically commercially available unit (such as a General Radio P-40). On the ordinate, the rms voltage output capability in dB re 1 volt is plotted. The 0 point corresponds to a sound pressure level of 134 dB when used with a microphone whose sensitivity is -60 dB re 1 volt/μbar. On the abscissa, the frequency times the cable length (assuming 50 pF/ft cable capacitance) is plotted. Thus, the abscissa corresponding to a 1000-foot cable driven with a 10 kHz signal is 10^7. The curve shows that the commercial unit is capable of less than 0.25 V rms of drive whereas the new unit is capable of almost 3.0 V rms. This new unit is battery-operated and requires very little idling current. Its efficiency is very high since it draws power only as required to drive the cable. When driving a 1000-foot cable terminated in a 10 K load, the amplifier is flat $\pm$ 0.5 dB from 1.3 Hz to 75 kHz. Its source impedance is 3 ohms in series with 180 mF and the output noise level with shorted input is 10.5 μV rms.

The high input impedance preamplifier required for good low frequency response from ceramic microphones and dc-biased condenser units is frequently affected by humidity. This is especially so when the amplifier is taken from a hot humid area to an air-conditioned repository. Moisture condenses on the outside of the amplifier and on the inside, as well, creating unwanted conduction paths in the amplifier which not only can lower the input impedance but can actually change the operating point of the input FET sufficiently to limit the dynamic range of the unit and even cause blocking. The problem can persist for quite a while after the unit is returned to the humid field. To avoid this, the critical circuitry and the input connector have been encapsulated with a rubber-like compound such as Sylgard.

L. RF Pickup

Another field problem associated with the use of long cables is RF pickup from nearby transmitters. Although the trouble can manifest itself in several ways, the typical problem is caused by excessive RF levels at the output of the preamplifier or line driver. Generally these units use extensive feedback from the output and RF signals are thus fed back to the input where they cause nonlinearities. The solution is

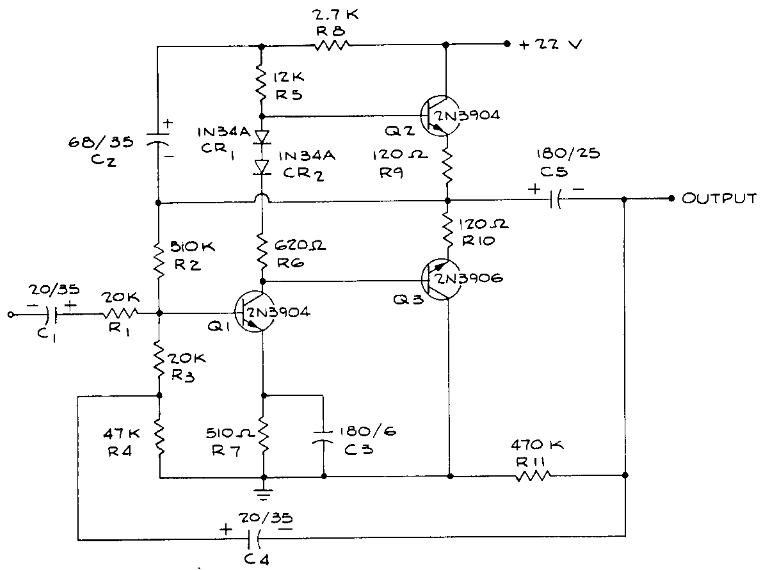


Figure 1-9. Line-Driving Amplifier Schematic

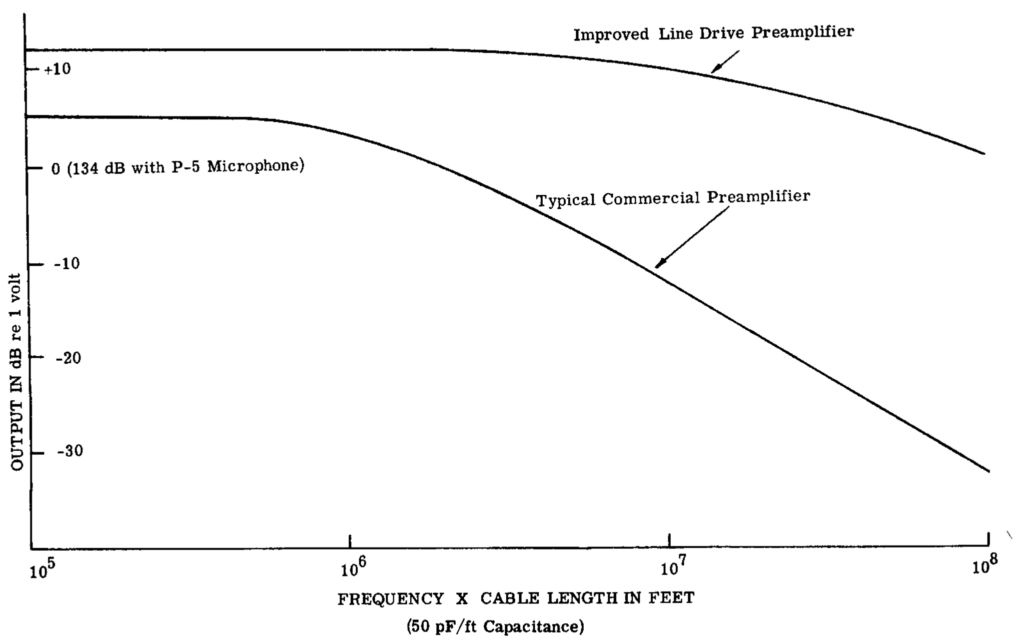


Figure 1-10. Line Driving Capability

to use RF traps at the output of the line driver. Careful attention must be paid to the quality and location of grounding points. Finally, all transmitting equipment should be kept as far as possible (at least 100 feet away) from the measuring microphones and the recording setups.

M. Electrolysis

When the preamplifier is powered by direct current through the same transmission line that carries the signal, one can experience under certain circumstances noise impulses which sound like crackling. This noise has been attributed to electrolyzation in the line and has been successfully eliminated by powering the preamplifiers from local batteries.

N. Controlling the Recording Level

The data collector is faced with the problem of accepting a wide range of acoustic signal levels--often as much as 80 dB or even more. Most data recorders whether operating in the direct or FM mode have signal-to-noise ratios of the order of 40 dB. Since it is normally necessary to stay at least 10 or 15 dB above the system noise, only 25 or 30 dB may be left for dynamic changes in the data: The input range must be accommodated by suitable gain changes in the recording system. At all times the accuracy of calibration must be maintained.

The solution adopted by the AWG has been to modify commercially available sound level meters; their accurate 10 dB step attenuators are used for gain adjustment. The sound level meter provides a convenient means for monitoring record level as well as for setting it properly. The main problem encountered is the limited dynamic range (16 dB) displayed on the meter face, which is due to the use of a suppressed-zero meter. In field operation, the data collector should record no higher than 10 dB below full level to accommodate peaks which are too short in duration to register on the meter. In many instances, an even larger safety margin is required. For small signal levels the meter barely deflects and the operator has no assurance that he is recording. Therefore, in the acoustical surveys which have provided a basis for this chapter, the meter has been modified to remove the suppressed zero and replace the face. With this modification (Figure 1-11), the entire 40 dB range of the tape recorder is shown. Since this modification was made, it has been found that very few changes in recording gain are required. Consequently, the data taken is much easier to reduce and the signal levels can be readily maintained in a good operating range of the recorder.

Figure 1-11. Modified Sound Level Meter

When recording impact data or other transient phenomena, it is helpful to have available a portable oscilloscope and a peak-storing meter such as an impact analyzer. This equipment will allow the operator to monitor the peak recording level even for signals whose duration is too short for the sound level meter's ballistics to follow. The multiple track recording technique using staggered recording levels may be the only solution to capture vital data which cannot be repeated and whose level is unknown in advance.

Amplifiers are available which sense the signal level and automatically change their gain in known controllable steps. Frequently, however, this automatic feature can hamper operations since the amplifier does not have the same knowledge of future events which is available to the human operator. Thus, gain changes may be more frequent than necessary with automatic equipment. Also, a separate track for recording gain changes usually is needed.

O. Tape Recorders

The choice of a tape recorder is critical in the acquisition of good data. The need for light weight and battery operation limits the choice of instruments available. Certain sacrifices in recording versatility and reliability have been made in the design of truly portable equipment. Both 1/4" and 1/2" IRIG format Lockheed Model 417 recorders have been used in the field with very good success. These units allow a choice of recording speeds to suit most acoustical data-gathering requirements. Frequently it is necessary to record in parallel using both the FM and Direct modes to cover the frequency range desired. For example, in the Direct mode, at 7-1/2 ips a typical frequency response would extend from 100 Hz to 25 kHz. To record lower frequencies, the FM capability is used to cover the range from DC to 2500 Hz.

The FM mode has accuracy advantages over the Direct mode in that recorder amplitude variations have a minimal effect. Whereas amplitude stability in the Direct mode cannot be guaranteed to better than approximately 1.0 dB, in the FM mode stability of 0.1 dB can be achieved. Furthermore, the phase characteristics are more readily controlled in FM than in Direct recording. This is especially important in situations in which wave shape, rather than mere frequency content, is important.

P. Recorder Calibrator

It is advantageous to make an electrical field calibration of the tape recorder to establish its response, noise level, and maximum recording level. The latter measurement allows one to rapidly set up the proper record level for duplication. Although a complete calibrated frequency sweep is ideal and should be used wherever possible, a square wave calibrator has been developed for situations in which a more rapid calibration is required. This unit produces signals of 25 Hz, 250 Hz, and 2500 Hz. The level of the signal is very stable and compatible with the tape recorder requirements. With this signal recorded on the tape, a means is provided in data reduction for ascertaining the entire frequency response of the recorder. By frequency analyzing the square wave signal at the data reduction center, the harmonic analysis produced can be compared with that of a true square wave, verifying the recorder performance. Seen in Figure 1-12, together with its schematic, the unit is small and rugged.

Q. Recording Tape

A high grade instrumentation tape should be used. A polyester backing is recommended. One-and-one-half-mil tape provides lower print-through and greater strength than one-mil tape, but the latter can be used where extra recording time per reel is desirable. Tape thinner than one mil should be avoided.

Extreme care should be taken to avoid contamination of the tape. It should be stored in a plastic bag and a covering box. One should not touch the recording surface of the tape, thus avoiding fingerprinting which causes dropouts. Sufficient leader and trailer should be left to allow tape threading without contamination of the data.

After recording, the tape should be protected from stray magnetic fields. Shielding containers are ideal for this.

1.5 FIELD PROCEDURES

A. General

The single most important factor in assuring the success of a data-gathering mission is adequate planning and implementation of field procedures. Regardless of the sophistication of the field equipment, success of the endeavor is jeopardized if sufficient effort is not devoted to this initial step. Conversely,

the suitability of the equipment for the task at hand is determined by the plan and if the planning is done sufficiently in advance of the test, as is proper, adequate equipment will be provided in time for the mission.

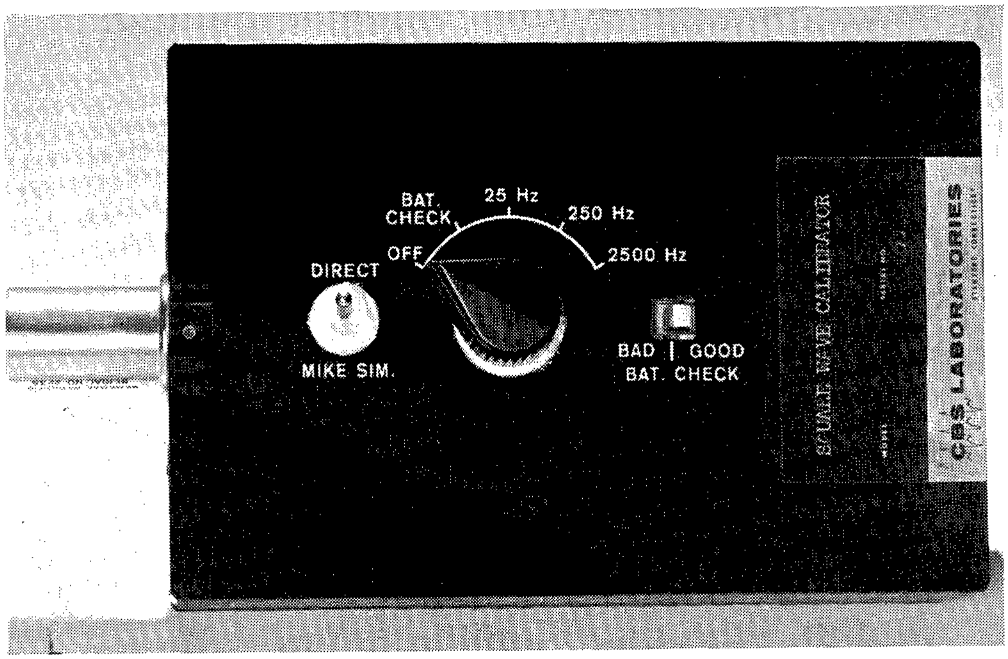
B. Development of the Test Plan

The key to the development of an adequate test plan is the foreknowledge of the use to which the data is to be put, i.e. the goals of the testing program. Each test must have its own purpose which must be clearly spelled out. The pressure of scheduling frequently tempts one to skimp on the identification of the purpose of each test. However, the time saved at the outset is most assuredly lost later in attempting to cull the wheat from the chaff of data taken.

For example, if the purpose of a specific test is to determine the effectiveness of a certain design change, say, that of a different windscreen on a device, the test plan must provide for sufficient controls on the experiment to separate definitively this change from all others. The plan must include provisions for the testing of Device A with the new windscreen simultaneously with identical Device B with the old windscreen. The windscreens should subsequently be interchanged to remove the effect of individual device characteristics. The devices should be checked during periods of wind and also in periods of calm. Obviously, wind velocity measurement should be recorded. The instrument used for this should be capable of measuring peak velocities during gusts rather than simple averages. If the device employs circuitry which changes its characteristics as a function of ambient noise, the tests must be run under various ambient conditions.

In short, a complete knowledge of the device to be tested and the results expected from the test are necessary in order to formulate a plan which takes into consideration all foreseeable contingencies and which assures that the causes and effects can be assigned without ambiguity. Thus, even if the organization which fields the data-gathering team is not also charged with the responsibility for preparing the test plan, it should participate in formulation of the plan from the very inception.

Once the purpose of the test has been defined, the field team leader can prepare the detailed test plan. The contents of the plan and the detail to which it is drawn up will vary depending upon the test to be performed, its location and its duration, etc. Typical examples of a Calibration Procedure and a Test Operation Procedure are shown in Appendixes A and B.



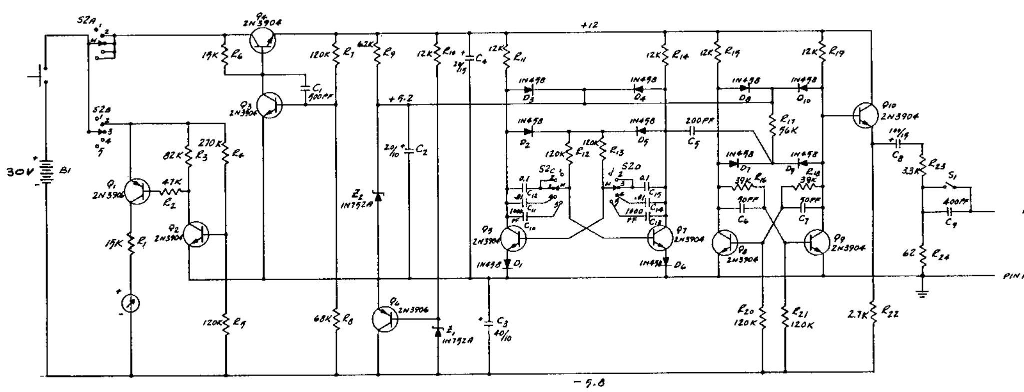


Figure 1-12. Square Wave Calibrator with Circuit Diagram

1.5.B

These were prepared for a specific one-day test during which outputs of two or three sensors were recorded at each of several sites. The Calibration Procedure was followed and performed the day before the actual test since COMEX ("Commence Exercise") was quite early (0600). In addition, on test day a brief calibration procedure was performed on all of the data gathering microphones.

The key portions of the Calibration Procedure are:

1. Voice annotation of:

   a. Test Designation and Classification of Tape
   b. Location of Test and Test Site
   c. Reel Number and Previous Reel Number
   d. Date
   e. Time
   f. Personnel at Site
   g. Recorder Identification
      (1) Make
      (2) Model
      (3) Serial Number
      (4) Recording Speed
      (5) Track Identification, i.e. which information is recorded on which track
   h. Site Description including Sensor Locations

2. Noise Level Measurements:

   a. Recorder noise alone, i.e. recorder input terminals shorted directly.
   b. System noise, i.e. data lines terminated at microphone end. This should be repeated for every record level setting which one anticipates using. The termination should be equivalent to the source impedance of the microphone.

3. Identification of Sensors:

   Each sensor is connected individually to the recorder and identified by speaking into it or by causing it to respond in some way. This gives a positive indication on the tape as to which sensor is which regardless of possible later discrepancies in tape markings or handling. Markings on the tape reel, alone, are insufficient since it is possible to transfer the tape

to a different reel which may have erroneous or inapplicable markings. While each sensor is being identified directly, a voice commentary is made to identify the sensor and the entire associated data-gathering chain.

4. Overall Acoustic Calibrations:

a. The data-gathering microphones are calibrated using a General Radio 1562A Sound Level Calibrator. This unit produces a precise 114 dB re 0.0002 dynes/ cm^2 sound pressure level at a variety of frequencies.

b. Other acoustic sensors are calibrated using a portable oscillator, battery-powered amplifier, and trumpet. Using this equipment, a sound field is set up in the vicinity of the item to be calibrated. The sound field is measured using a calibrated sound level meter. The field is probed all around the item to assure uniformity of field. Annotation is made of the sound pressure level used.

c. In both cases above, the setting of the recording level control, i.e. the sound level meter, is annotated.

5. Derived Calibrations:

In some cases it is not possible to put a primary acoustic calibration signal on each reel of tape. For example, in the particular case for which the procedures illustrated here were written, the test could not be halted once underway. Thus, it was necessary to establish derived calibrations on each reel of tape. These derived calibrations can be related in level to the overall acoustic signals. Therefore, on the calibration tape, in addition to the overall acoustic signal, a 250 Hz square wave (from the square wave calibrator) was recorded. (To cover a broader range, 25 Hz and 2500 Hz wave also could have been recorded.) This same signal was subsequently recorded on each reel of tape when it was not possible to perform the acoustic calibration. By once establishing the relationship between square wave level and acoustic signal level from the calibration tape, one can derive acoustic signal levels for all subsequent tapes. The square wave signal has another purpose in that a spectrum analysis of the signal when compared with the theoretical relationship between the harmonics of a square wave will indicate the frequency response of the data recording system.

1.5.B

6. Voice Annotation of the calibration procedure which was used and the method which is to be followed in data reduction to establish true sound pressure levels.

A Test Operation Procedure is prepared for the actual test, itself. It is invaluable in assuring uniformity in methodology and also serves as a checklist of things to do on each reel of tape. The key portions of the procedure are:

1. Voice Annotation of:

   a. Test Designation and Classification of Tape
   b. Purpose of the Test
   c. Identification of Reel Number in the Series
   d. Location of Test and Test Site
   e. Date
   f. Time
   g. Recorder Identification
      (1) Make
      (2) Model
      (3) Serial Number
      (4) Recording Speed
      (5) Track Identification
   h. Sensor Location, i.e. Height, etc.

2. Secondary Calibration using the Square Wave Calibrator track-by-track

3. Identification of Sensors:

   Each sensor is connected individually to the recorder and identified as to location, etc. by speaking into it or causing it to respond in some way. This gives a positive identification on the tape as to which sensor is which regardless of possible later discrepancies in tape markings or handling. Markings on the tape reel alone are insufficient since it is possible to transfer the tape to a different reel which may have erroneous or inapplicable markings. While each sensor is being identified directly, a voice commentary is made to identify the sensor and the entire associated data-gathering chain.

4. Overall Acoustic Calibrations:

   Same as Step 4 in the Calibration Procedure.

5. Annotation:

    a. Personnel at Site
    b. Description of Site
        (1) Terrain
        (2) Foliage
        (3) Local anomalies
        (4) Weather conditions
        (5) Other pertinent data

6. At this point the site is ready for data gathering and informs Central Control of the fact.

7. This procedure should be followed for each reel of tape taken during the test. If this is not possible due to time limitations, at least the first two steps must be performed for each reel. The entire procedure is then performed once at the beginning of the day's testing and again at the end.

C. Equipment and Personnel Requirements

During the planning phase of the test, after the purpose has been established, the equipment required for the exercise is determined and made ready for the field. An Equipment Checklist (shown in Appendix C) is made up for each site. Using it assures that missing, defective, or insufficient equipment is recognized at an early stage in the mission in time for remedial gathering missions. Special requirements arise on occasion which are not included on the general list. These are added as needed and the list revised. When the time arrives for deployment to the sites, each site captain uses the checklist to load his equipment. If this list has been properly prepared, each site should be self-sufficient for ordinary operation and also for those contingencies which have been anticipated by experience and careful preplanning.

At this point, it is time to determine the personnel requirements including any local assistance that may be available. The best approach is to list the tasks and assign personnel and dates and times to each task. For the actual test, itself, three persons per site are desirable. Occasionally four are required and sometimes two per site are sufficient. In general, one person is needed for every four tracks of recording to handle the monitoring and the recording level adjustments. If the data to be recorded will necessitate frequent and/or unforewarned changes in recording level (such as would be the case with unscripted aircraft tests), it is best to plan on one man for every two

tracks of data. One man is required to keep the written detailed field log (Appendix D). If the location and testing require the use of a "spotter," another man must be added to the roster.

Regardless of other considerations, safety dictates that a single man should never be left in the field alone. Under difficult jungle conditions, two men are needed to calibrate the microphones, which means a minimum of three people per site required. However, some of the tasks, such as keeping the field log and spotting do not require technically trained personnel, and it may be possible to use such local assistance as might be available at a military base.

The above requirements are based on a test of relatively short duration. If the test is planned to continue for more than 12 or 14 hours on a single day, or will exceed an 8-hour-per-day-average over several days, backup teams should be provided. This is especially the case if the tests are performed under severe conditions such as in a jungle environment in the rainy season. If the tests are to continue over a period of several months, the teams should be rotated at reasonable intervals, typically every two to four weeks depending upon field conditions. It must be remembered that situations change rapidly during tests and the field team is called upon to make rapid and accurate decisions concerning the conduct of the test, operation of the equipment, and performance of the measurements. If the personnel are strained beyond reasonable capacity, their alertness will suffer affecting their accuracy and efficiency.

D. Scheduling the Tasks

All of the above planning and preparation should be done before deployment to the field. This presupposes a sufficient period of time between the initiation of preparations and the test itself. The time required for adequate preparation will, of course, depend upon the type of test involved, similarity to past tests, and expected duration. In general, however, at least a two-week period should be set aside for preparation. If an entirely new field of exploration is to be opened, a period of four weeks or even longer should be planned for adequate organization.

Based upon the test plan which has been drawn up, a schedule of events should be prepared. Designating the day of commencement of the test (COMEX) X-Day, a dry run including system calibration should be performed on day X-minus-1. It is preferable to have all work finished by noon of day X-1.

A last-minute briefing should be held early that afternoon and the rest of the day devoted to laying out the cables and testing them for continuity, RF pickup, noise, etc. If there are extensive amounts of cable to be laid, or if the terrain is especially severe, one or two extra days should be set aside for this task. On day X-2 the equipment should be packed according to site and in accordance with the checklist prepared previously. It will then be ready for deployment on day X-1. Previously arranged suitable security measures for the equipment should be placed in effect by noon of day X-1. The sites for the sensors should be selected the day before beginning laying of the cable, i.e. day X-3 or sooner. A check that the cable has been properly laid should be made the afternoon of day X-2.

In locating the sites, due care must be taken of the acoustic propagation expected between the sound sources and the sensors. For example, a different arrangement may be needed for studying propagation loss than when accumulating signature data on vehicular noise. The trees or locations for hanging the sensors can be blazed with suitable cloth or string. At the time the sites are planned, a map of each site should be prepared. The distances from the sensors to benchmarks which can be found on master maps should be measured and noted. A vertical diagram of sensor deployment should be made, and Polaroid pictures of the terrain should be taken.

In addition to the site preparation, there are many supporting operations which must be arranged in the days before COMEX. These include communications between members of each site's data-gathering team, and between each site and a central clearing headquarters. Further, transportation to the sites must be provided as well as food, shelter, and suitable clothing for the personnel involved in the site activities. Shelter may be in the form of a tent, van, or station wagon large enough to house the equipment. Both the men and equipment must be adequately shielded from sun and rain. The logistics of transporting the men to the various sites on the day of the test must be carefully studied in advance. In general, a strict schedule must be adhered to and planning and teamwork are of the essence.

E. Implementing the Plan

During the afternoon of day X-1, a briefing should be held to alert the team to any last-minute changes, and to go over the test script if one is available, or, if not, to discuss

the types of signals which are likely to be encountered during the test. Depending upon the types of signals to be recorded, the leader will instruct the team concerning the best recording level to be used. For example, if impulsive sounds are expected, or if the sound pressure level is expected to rise abruptly without warning, he may instruct the team to leave 20-30 dB of headroom in setting ambient recording levels. On the other hand, if the recording is of a slowly varying signal such as the ambient, he may instruct them to leave only 10 dB headroom. He may alert the team to any special logging requirements which may be appropriate to the particular data to be taken. In general, this is the time again to go over any deviations from the normal procedure.

On X-day, the teams should be transported to the field and in position at least one hour before COMEX. If possible, a two-hour period should be prescribed. This gives the team a reasonable opportunity to calibrate the equipment properly, make appropriate annotations, and call for replacements on any equipment which might prove defective at the very last moment. Any "extra" time will be useful for preparing labels to go onto the reels of tape as they are recorded. During the actual test the field operators will be kept extremely busy setting recording levels, monitoring the quality of the recordings aurally, and keeping correct logs.

Besides logs of events as they occur, the teams should keep logs of equipment failure. In the severe environment and with the handling under which the data-gathering equipment operates, failure rates are apt to be quite high. It is important, therefore, to know what type of equipment is failing and in what way it is proving defective so that appropriate steps may be taken to improve its reliability in the future.

As soon as possible after the test, a debriefing should be held by the team leader to cull the information required for an accurate trip report. This session should cover any difficulties each team encountered, the steps taken to remedy them, any deviations from the plan which were necessitated, comments as to the quality of the recordings, suggested ways of improving the quality of the recordings on future tests of a similar nature, etc.

On the day of the test, the team leader should be in ready communication with the site teams and with the exercise directors. Decisions which cannot be made at the sites by the team members are referred to the team leader, who may have to visit

the particular site to investigate a problem at hand, deliver back-up equipment, etc. At the end of each day's exercise(s) all of the recordings are delivered to the team leader. Prior to X-Day, arrangements should have been made concerning the disposition of the recordings such as storage and/or transportation to their final destination, with special care given to the handling of any classified material. The tapes should be protected preferably with shielding cans to prevent accidental demagnetization. As an emergency measure, a few inches of air space (at least four inches) around the tapes will help if cans are not available and the stray fields are low.

From the timetable outlined above, it becomes clear that the field team should be at the testing area from three to seven days in advance of COMEX, depending upon the extensiveness of the testing and the advance preparation which might be required.

F. Communications

Second in importance only to adequate planning, is good field communications. Regardless of the high degree of prior planning and scripting, it is virtually impossible for the test to go exactly according to plan. For this reason alone, communications between the field teams and headquarters are essential if successful signal gathering is to be accomplished. Communication by field radio is convenient, however, a fairly large number of radios are required and may not be available at a particular site. Further, there may be a question of compromising security when the radio is used. In such a case, the answer may be wired field telephones. If these are to be employed, sufficient lead time must be left to allow communications lines to be strung.

In addition to requiring communications to accommodate changes in program plan, the field team may need a virtually continual communication link for certain types of tests. For example, if the purpose of the test is to record overflights of aircraft in a dense jungle, it may be necessary to give each team notice that an overflight is imminent just prior to the action so that they can adjust recording levels and identify the particular aircraft. Unless so alerted, it is sometimes difficult to tell that an aircraft is approaching until it is practically directly overhead. By this time the tape is severely overloaded and the data has been lost. However, such continuous notification can put a severe strain on the communication link if it is expected to serve other purposes as well. Thus, it may be necessary to allocate a

communications link exclusively to the data gathering teams. Alternately, spotters for each team can be provided with field telephone communications at their site. Whichever form it takes, adequate communications must be furnished to permit success of the data gathering mission.

1.6 FIELD LABORATORY PROCEDURES

A field laboratory is provided for the checkout and maintenance of the field equipment. Even for the shortest of tests, it is good practice to thoroughly test the data gathering system before deployment to the field since calibration can be upset during transportation from the home base. The need for a field laboratory is even greater when the testing program extends over a considerable period of time. Here, the laboratory should be suitably equipped for repair and maintenance of equipment as well as checkout.

In addition to initial checkout prior to the first day of testing, all equipment should be cleaned and tested as soon as possible after the exercise. During lengthy data-gathering missions, double sets of equipment may have to be provided so that one set may undergo maintenance procedures while the other is in use. Additional people may also be required to perform the procedures.

In Appendix E, calibration procedures are given for the sound level meter, preamplifier and microphone. Besides maintenance of this equipment, the most critical function of the field laboratory is the maintenance of the tape recorders. Being an electromechanical device, the tape recorder is exceptionally prone to malfunction in the field. It must be scrupulously cleaned, heads must be demagnetized, and mechanical components such as drive belts checked and periodically replaced. Besides this mechanical upkeep, the tape recorder's sophisticated electronics design necessitates a careful checkout of its record and reproduce electronics. To fully check out and adjust a tape recorder and associated field equipment requires at least a full eight-hour day.

In addition to maintenance of equipment, battery storage and recharging facilities are maintained in the field laboratory. The recharging cycle for a nickel-cadmium battery is 13 hours. It can be seen that several sets of batteries and recharging facilities are required if this type of battery is used.

When accumulating data in a very humid environment, a desic-

cator and a supply of disiccant should be included in the field laboratory equipment. This can be used to dry out microphones and high impedance circuits such as are found in the preamplifiers and sound level meters.

Especially, during long data-gathering missions it is advisable to provide the field laboratory with sufficient data reduction equipment to take a "quick look" at the data being received from the field. On the basis of this preliminary reduction, plans can be modified or key aspects of the test repeated as required to assure accurate and reliable data.

1.7 WRITTEN FIELD PROCEDURES

Representative field procedures are appended as follows:

Appendix A:	Field Calibration Procedure
Appendix B:	Field Test Operation Procedure
Appendix C:	Field Equipment Check List
Appendix D:	Field Log
Appendix E(1):	Before-Use Equipment Checkout and Calibration for Sound Level Meter
Appendix E(2):	Before-Use Equipment Checkout and Calibration for Preamplifier
Appendix E(3):	Before-Use Equipment Checkout and Calibration for Microphone

CHAPTER TWO

DATA ANALYSIS AND REDUCTION

CHAPTER TWO: DATA ANALYSIS AND REDUCTION

2.1 INTRODUCTION

A. Purpose

The purpose of this chapter is twofold: (1) to discuss in broad terms the appropriate factors to be considered in data reduction and analysis; and (2) to describe, with examples, many practical ways in thich data can be manipulated into formats useful to an analyst. While the mathematical relationships for the various data transformations are stated, it is not the purpose of this chapter to validate these relationships. Such validations are available in many standard texts, some of which are referenced. Finally, while most of the discussion in this chapter is oriented to audio-frequency data, with appropriate scaling factors the techniques are equally adaptable to seismic, infrasonic, ultrasonic, and similar phenomena.

B. General Aspects of Data Reduction and Analysis

Data acquisition, reduction, and analysis should be viewed as an integrated process. One begins with the establishment of realistic objectives and the formulation of hypotheses and plans respecting the type of data acquisition and reduction that will be required as described in chapter 1. Such preparatory work enables the analyst to proceed with dispatch to the task of processing the data as soon as it becomes available. The results of an acoustical data-acquisition mission are usually contained in one or more reels of tape, supplemented by plans, logs and debriefings of the data-acquisition teams.[1] The actual task of data reduction and analysis can be said to begin at this point.

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[1]See Chapter 1, pp. 1-23 to 1-34.

2.1.B

Almost invariably, the first task of the analyst is to perform a preliminary overall review of the newly acquired data, to classify it in accordance with various categories and priorities, and then to proceed to "reduce it," that is, to transform it into compact and manageable formats in order to extract from it the needed information. The task of categorizing and evaluating data requires considerable experience and judgment which cannot be spelled out in procedural detail. The general principles, however, are summarized in subsection 2.1.C. The successful outcome of a data mission is often ensured by proper utilization of data in these various categories.

Because the results of analysis often lead to the design of electroacoustical or electronic devices, the data is usually reduced in terms of its frequency and/or time-domain attributes, which provide essential information to the equipment designer. A considerable portion of this chapter is devoted to the various frequency and time domain data-reduction procedures that have been applied successfully to the tasks of the Acoustical Working Group (AWG).

Often, the results of frequency domain and time-domain analysis suggest the necessity of employing special data-reduction techniques or apparatus for the detection and acquisition of the targets of interest against background noise or other interfering phenomena. Cogent among these are the temporal correlation of signals received by two or more spaced-apart transducers; or comparison of signals received by coincident transducers with differential directional characteristics; or the use of special directional arrays which augment the S/N in certain directions; and even demodulation methods which reveal the existence of sonic phenomena which might not have been acquired with conventionally available acoustic detectors. The varieties and configurations of such special techniques and devices are limited only by the skill and inventiveness of the analyst and the designer; some of these devised during AWG tasks are outlined in Section 2.6.

It should be noted that the terms "reduction" and "analysis" sometimes have been used interchangeably. The term "frequency analysis" is deeply ingrained in the language of the mathematician to denote reduction in the frequency domain, and we have respected this tradition in the preparation of this chapter. The term "data reduction" usually involves the process of formulating or plotting mathematical concepts in the time domain. For example, a time series representing sound-pressure level

may be "reduced" in terms of its peak-level envelope as a function of time, or in terms of intensity variation of its frequency attributes as a function of time. As a last step after data reduction, data analysis is employed to apply these results to the solution of the problem at hand.

A useful tool for the analyst and the designer is a mathematical model of a signal, or a background noise, or a data-acquisition detector, etc., which allows the performance of the system to be analyzed by means of a computer. In a text of this nature, the complete range of modeling techniques cannot be fully explored; however, models of the simplest concepts employed in the work of the AWG are given toward the end of the chapter.

C. Categorization of Data

1. Primary Data

Data reduction, as defined in the previous subsection, is a process of transforming raw data into other formats, the purpose being to produce formats that are more easily related to objectives than are the original data. To begin with, let us consider the primary data. Primary data is the physical data directly related to the objective. It is that data which is reduced and analyzed to achieve the objective.

Two general types of data are processed: time-ordered data and time-random data. For the most part, the AWG-acquired data was time ordered. The distinction between the two types is that for the time-ordered data there is useful information in the exact sequence of values encountered, whereas for time-random data there is no such useful information.

To illustrate time-ordered data: the sound output of a gassoline engine exhibits, with time, certain interesting cyclical sound outputs that would be destroyed if one were to destroy the sequence of instantaneous sound outputs; at the same time, however, the frequencies of the major cyclical components are not constant, but instead wander in some random fashion. In contrast to this type of data, if one were to measure the detection ranges of an acoustic-detection system working against a given target, the sequence of ranges obtained for sequential target runs would constitute time-random data. No information would be destroyed if the sequence of ranges were changed.

The reductions (transformations) for time-ordered data should preserve pertinent aspects of the original sequencing

information, since such information in these signals is useful. There are two general types of transformation possible: (1) a transformation that destroys no information in the original--this is a one-to-one type of transformation; and (2) a transformation that destroys information in the original data--this is a many-to-one type of transformation. In the first type of transformation, one can combine the transformed data to reconstruct the original data in its entirety. The Fourier Transform (see subsection 2.2.C.) is theoretically a one-to-one transformation since a given set of Fourier components, suitably combined, produces a unique original signal. In practice, this is only approximately true, since one would have to take an infinitely long data sample and reconstruct it with an infinite set of oscillators to exactly transform the data. Inasmuch as one customarily selects only a segment of data for analysis, an exact transformation is not obtained.

In the second type of transformation, one cannot combine transformed data to produce a unique original signal. The power-spectral-density transformation is an example of the second kind of reduction. A given power-spectral density does not correspond to a unique original signal, but instead pertains to many.

Signals to be reduced may contain unique time-ordered information in many aspects, either of waveform shape or of waveform parameters derived from the signals. Waveform shapes involve sequential details in the instantaneous signal values, or in composites of these details, like envelopes. Time-domain reduction is often used to process waveform-shaped signal aspects. Waveform parameters are items such as peak-to-average ratios, rms values, phases, etc. Information-destroying reductions highlight particular signal aspects by eliminating extraneous aspects, thereby allowing one to focus attention more easily on the significant features.

2. Calibration Data

Calibration information is required to ascertain absolute dimensions of data. For example, suppose one requires that the voltage analogs of sound recorded on magnetic tape be scaled in terms of sound pressure level. This is made possible, in convenient form, by recording a signal of known sound pressure on each reel of tape to be analyzed.[2]

System performance validation data is a part of calibration information. It is introduced into test procedures to ensure that the data characteristics are valid and not due to the particular data acquisition and reduction techniques. Good vali-

[2]See Chapter 1.

dation data is absolutely necessary to resolve anomalies arising in data reduction.

The validation data contains test data on the acquisition system to determine its limitations and usefulness. Such data is usually acquired in the laboratory both prior and subsequent to the field tests, and also in the field during the test. Among other things, this data includes such items as acquisition system bandwidth, dynamic range, noise level and frequency stability. The data should validate the fact that established system performance has not changed during the test, or it should provide the necessary correction information for those system changes which do occur.

3. Test Procedure Data

Test procedure data is taken to provide the means for proper interpretation of the results of data reduction. Without good test procedure data, one cannot recheck, interpret, or extrapolate the results of a given test.

The test procedure data includes the detailed test plan and the time correlation of the test events with other test data. To illustrate, if certain targets are run, information should be recorded on model number, speed, altitudes and other aspects that would uniquely characterize the target. Information on environment, acquisition system settings, unusual test factors, for example, are also of importance during the reduction. The absence of these procedure data limits, at best, the use of good primary data and, at worst, renders it useless.

The process of categorization of data as outlined above assists the analyst with the planning of the data reduction tasks and ensures that the reduced data can be reliably applied to the solution of mission objectives.

D. Definitions and Units

To avoid misinterpretations, the following terminology is used hereafter. The reader is referred to ANSI Standard S1.1-1960 for a complete listing of acoustical terminology and definitions.

Periodic Quantity. A periodic quantity is an oscillating quantity whose values recur for certain increments of the independent variable.

Note 1: If a periodic quantity v is a function of t, then

$$v = F(t) = F(t + T)$$

where T, a constant, is a period of v.

2.1.D

Primitive Period (Period). The primitive period of a periodic quantity is the smallest increment of the independent variable for which the function repeats itself.

Note 1: If no ambiguity is likely, the primitive period is simply called the period of the function.

Frequency. The frequency of a function periodic in time is the reciprocal of the primitive period. The unit is the Hertz (Hz).

Angular Frequency (Circular Frequency). The angular frequency of a periodic quantity, in radians per unit time, is the frequency multiplied by 2π. The usual symbol is ω.

Peak-to-Peak Value. The peak-to-peak value of an oscillating quantity is the algebraic difference between the extremes of the quantity.

Phase of a Periodic Quantity. The phase of a periodic quantity, for a particular value of the independent variable, is the fractional part of a period through which the independent variable has advanced, measured from an arbitrary reference.

Harmonic. A harmonic is a sinusoidal quantity having a frequency that is an integral multiple of the frequency of a periodic quantity to which it is related.

Ambient Noise. Ambient noise is the all-encompassing noise associated with a given environment, being usually a composite of sounds from many sources near and far.

Random Noise. Random noise is an oscillation whose instantaneous magnitude is not specified for any given instant of time. The instantaneous magnitudes of a random noise are specified only by probability distribution functions giving the fraction of the total time that the magnitude, or some sequence of magnitudes, lies within a specified range.

<u>Spectrum.</u> The spectrum of a function of time is a description of its resolution into components, each of different frequency and (usually) different amplitude and phase.

<u>Spectrum Density (Power Spectrum).</u> The spectrum density of an oscillation is the mean-square amplitude of the output of an ideal filter with unity gain responding to the oscillation, per unit bandwidth; i.e., the limit for vanishingly small bandwidth of the quotient of the mean-square amplitude divided by the bandwidth.

<u>Line Spectrum.</u> A line spectrum is a spectrum whose components occur at a number of discrete frequencies.

<u>Continuous Spectrum.</u> A continuous spectrum is the spectrum of a wave the components of which are continuously distributed over a frequency range.

<u>Microbar, Dyne per Square Centimeter.</u> A microbar is a unit of pressure commonly used in acoustics. One microbar is equal to 1 dyne per square centimeter.

<u>Sound Pressure.</u> The sound pressure at a point is the total instantaneous pressure at that point in the presence of a sound wave minus the static pressure at that point.

<u>Maximum Sound Pressure.</u> The maximum sound pressure of any given cycle of a periodic wave is the maximum absolute value of the instantaneous sound pressure occurring during that cycle.

<u>Peak Sound Pressure.</u> The peak sound pressure for any specified time interval is the maximum absolute value of the instantaneous sound pressure in that interval.

<u>Effective Sound Pressure (Root-Mean-Square Sound Pressure).</u> The effective sound pressure at a point is the root-mean-square value of the instantaneous sound pressures, over a time interval at the point under consideration. In the case of periodic sound pressures, the interval

must be an integral number of periods or an interval that is long compared to a period. In the case of nonperiodic sound pressures, the interval should be long enough to make the value obtained essentially independent of small changes in the length of the interval.

Sound Energy. The sound energy of a given part of a medium is the total energy in this part of the medium minus the energy which would exist in the same part of the medium with no sound waves present.

Sound Intensity (Sound-Energy Flux Density) (Sound-Power Density). The sound intensity in a specified direction at a point is the average rate of sound energy transmitted in the specified direction through a unit area normal to this direction at the point considered.

Level. In acoustics, the level of a quantity is the logarithm of the ratio of that quantity to a reference quantity of the same kind. The base of the logarithm, the reference quantity, and the kind of level must be specified.

Sound Pressure Level. The sound pressure level, in decibels, of a sound is 20 times the logarithm to the base 10 of the ratio of the pressure of this sound to the reference pressure. The reference pressure shall be explicitly stated.

Note 1: The following reference pressures are in common use:

(a) 2×10^{-4} microbar

(b) 1 microbar

Reference pressure (a) is in general use for measurements concerned with hearing and with sound in air and liquids, while (b) has gained widespread acceptance for calibration of transducers and various kinds of sound measurements in liquids.

Band Pressure Level. The band pressure level of a sound for a specified frequency band is the sound pressure level for the sound contained within the restricted band. The reference pressure must be specified.

Note: The band may be specified by its lower and upper cutoff frequencies, or by its geometric

center frequency and bandwidth. The width of the band may be indicated by a prefatory modifier; e.g., octave band (sound pressure) level, half-octave band level, third-octave band level, 50-Hz band level.

Spectrum Level (Spectrum Density Level). The spectrum level of a specified signal at a particular frequency is the level of that part of the signal contained within a band 1 Hz wide, centered at the particular frequency. Ordinarily this has significance only for a signal having a continuous distribution of components within the frequency range under consideration. The words "spectrum level" cannot be used alone but must appear in combination with a prefatory modifier, e.g., pressure, velocity, voltage.

Sound Level. Sound level is a weighted sound pressure level, obtained by the use of metering characteristics and the weightings A, B, or C specified in American Standard Sound Level Meters for Measurement of Noise and Other Sounds, Z24.3-1944. The weighting employed must always be stated. The reference pressure is 0.0002 microbar.

Transmission Loss. Transmission loss is the reduction in the magnitude of some characteristic of a signal, between two stated points in a transmission system.

Microphone. A microphone is an electroacoustic transducer that responds to sound waves and delivers essentially equivalent electric waves.

Pressure Microphone. A pressure microphone is a microphone in which the electric output substantially corresponds to the instantaneous sound pressure of the impressed sound wave.

Directional Microphone. A directional microphone is a microphone the response of which varies significantly with the direction of sound incidence.

Directional Response Pattern (Beam Pattern). The directional response pattern of a transducer used for sound emission or reception is a description, often presented graphically, of the response of the transducer as a function of the direction of the transmitted or incident sound waves in a specified plane and at a specified frequency.

2.2.A

2.2 THEORY AND PRACTICE OF DATA REDUCTION AND ANALYSIS

A. Introduction

While man lives in a time-domain world, the recognition of many natural and man-made sounds is based upon the repetitive aspects of the time series. Such natural manifestations can be identified in the song of a bird, in the cry of an animal, or in the aeolian tones caused by wind passing through the branches of a tree. Most man-made sounds such as the cyclic repetitiveness of a truck engine, the whine of a saw mill and the humming of power lines can be usefully and conveniently recognized in terms of repetition rate, or in the frequency of energy interchange between the potential and kinetic states.

Data reduction consists of the extraction of salient features from the raw data. There are two principal types of data reduction--reductions in the frequency domain (usually called frequency analysis) and reductions in the time domain (to be discussed in more detail in the following sections). From the utilization of both types of data reduction, information is gained about the phenomenon which allows it to be more fully characterized and understood. As a result of this data reduction, identification and categorization of the occurrence from its acoustic signature can be made.

By transforming the data into the frequency domain, using the techniques described in subsections 2.2.B, C, D and E, one can identify the regions of maximum energy concentration. Often, a distinctive "signature" which is characteristic of only one type of phenomenon is evident. In this case, the source of the data can be identified. In other cases, it is impossible to determine the precise origin of the data and yet, through frequency analysis, the most appropriate bandpass characteristics for sensor equipment can be determined. This allows optimum reception of one type of data with rejection of other types. By identifying the desired type of signal and the undesired types as noise, one can improve the signal-to-noise ratio of the sensor system.

Specific types of occurrences can often be identified by their signatures in the time domain as well. The rate of increase and/or decrease in signal level, whether it be of the instantaneous signal itself or the signal envelope, is often a clue to the identification of the signal. Similarly, the rates of change of signal in the time domain can be used to design circuits which will separate the desired signal from the surrounding noise. As a result of time-domain analysis, automatic

gain-control constants as well as post-detection filtering time constants can be determined.

Whether it be in the time or frequency domains, the techniques of data reduction in both their theoretical and practical aspects will be covered in the following sections. Very frequently, the methods used to reduce the data have serious effects upon the data itself and the conclusions to be drawn from the reduction. These occurrences will be discussed, as well as the interpretations which should be applied to the results of data reduction.

B. Fourier Series[3]

In general, periodic data stemming from natural occurrences can be analyzed into a sequence of harmonically related frequency components. The data which naturally occurs as a time series e.g., sound pressure as a function of time, t, can thus be converted into a frequency series, e.g., sound-pressure level as a function of frequency. Such a transformation from the time domain into the frequency domain is accomplished by Fourier Analysis. Assuming that the time series is deterministic, i.e., that its behavior is predictable in the form of a mathematical equation and that it is periodic (the function repeats identically and regularly every T seconds), the transformation into the frequency domain yields a discrete series of components at a fundamental frequency, f_o, equal to 1/T and harmonics thereof. The phenomenon can therefore be completely specified for all time by a series of the form

$$X(t) = \frac{a_o}{2} + \sum_{n=1}^{\infty} [a_n \cos 2\pi n(t/T) + b_n \sin 2\pi n(t/T)] \qquad (2\text{-}1)$$

where the Fourier coefficients a_n and b_n are given by

$$a_n = \frac{2}{T}\int_o^T X(t) \cos 2\pi n(t/T)dt \qquad n = 0, 1, 2, \ldots.. \quad (2\text{-}2)$$

and

$$b_n = \frac{2}{T}\int_o^T X(t) \sin 2\pi n(t/T)dt \qquad n = 1, 2, 3, \ldots.. \quad (2\text{-}3)$$

where $2\pi/T$ is the fundamental angular frequency, ω_o and where the angular frequency of harmonic components, ω_n, is $2\pi n/T$.

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[3]See, for example, Mischa Schwartz, Information Transmission, Modulation and Noise, McGraw-Hill Book Company, Inc., 1959.

2.2.B

Alternately, the Fourier Series given in Eq. (2-1) can be rewritten as

$$X(t) = c_o + \sum_{n=1}^{\infty} c_n \cos\,[2\pi n(t/T) - \theta_n] \tag{2-4}$$

where

$$c_o = a_o/2$$

and

$$c_n = \sqrt{a_n^{\;2} + b_n^{\;2}} \qquad n = 1, 2, 3, \ldots\ldots \tag{2-5}$$

$$\theta_n = \tan^{-1}(b_n/a_n) \qquad n = 1, 2, 3, \ldots\ldots \tag{2-6}$$

from which we can say that any periodic time series can be expressed as the summation of a steady-state or d-c term (c_o) and harmonically related oscillators of frequency $f_n = n/T$, amplitude c_n and relative phase angle θ_n. An alternate form of the Fourier series, which is frequently used, is in exponential notation. It can be shown readily that Eq. (2-1) is equivalent to

$$X(t) = \sum_{n=-\infty}^{\infty} d_n \exp[j2\pi n(t/T)] \tag{2-7}$$

where

$$d_n = \frac{1}{T}\int_o^T X(t)\exp[-j2\pi n(t/T)]dt \tag{2-8}$$

Data which can be transformed identically into a Fourier Series are called Periodic Quantity. (Simple Periodic Quantity refers to a pure sinusoidal signal--a rare occurrence in nature.) An example of this type of data is the sound or vibration from a reciprocating engine. Assuming that the engine has been operating from all time and will continue to operate for all time--an impossible but nevertheless practical assumption for any reasonable data period--the data can be identically transformed from the time to the frequency domain by a Fourier Series since all the data is harmonically related.

Data which stems from multiple sources may not be complex periodic even if the signals from each of the sources is complex periodic. For example, consider two nonsynchronized reciprocating engines such as on an aircraft. Each engine produces data which

is complex periodic. However, if the ratio of the fundamental periods is an irrational number, the fundamental period of the "beats" will not be a subharmonic of the fundamental period of the engine, and so a Fourier Series cannot be built on the beat frequency as a fundamental. Such data is called "almost-periodic." Although it is not possible to completely describe such data with a Fourier Series, useful information on the relative strengths of the signal lines can be obtained. That is, it is possible and useful to describe the data as a series of lines at nonharmonically related frequencies. However, it will not be possible to establish the phase relationships of the various components with respect to the time reference and so it is not possible to return to the time domain from the frequency domain.

C. Fourier Integral

As set forth in the previous section, data stemming from periodic occurrences can be transformed from the time domain to the frequency domain by expansion into a Fourier series based upon the reciprocal of the period as the fundamental frequency. This series is made of discrete lines at the fundamental frequency and its harmonics. All of the signal energy is concentrated at these discrete frequencies, and usually there are only a finite number of them. Thus, one can say that $c_1^2/\Sigma c_n^2$ percent of the total energy is at the fundamental frequency, $c_2^2/\Sigma c_n^2$ percent at the second harmonic, $c_3^2/\Sigma c_n^2$ percent at the third harmonic, etc.[4]

Frequently, the data to be analyzed stems from a transient phenomenon[5] which has a finite duration and does not repeat it-

[4] The commonly used term "% n-th harmonic distortion" refers to the % amplitude, and not the % total energy of the complex signal, and it is equal to the square root of the percentages of the total energy as defined in this paragraph.

[5] Actually, since no phenomenon has continued from $t = -\infty$ and will continue to $t = +\infty$, all natural data is "transient." However, if the data has existed and will exist for many periods, the assumption of infinite duration is justified for all practical purposes, and the Fourier series is applicable. The non-infinite duration reflects itself as a "broadening" of each spectral line into a $\frac{\sin \omega}{\omega}$ pulse. The width of the "broadening" is inversely proportional to the duration of the phenomenon. Thus, if the phenomenon occurs for an arbitrarily large number of periods, the width of the line and the subsequent error can be made arbitrarily small.

self, even if within the event there may be oscillatory functions. Since there is no period to form the basis of the series, one cannot expand the data into a discrete Fourier series. Examples of clearly transient phenomena are the backfire of an engine (considered by itself), the report of a gun, or a supersonic boom. The amplitude-time history of these phenomena can nevertheless be transformed into the amplitude-frequency domain by the Fourier Integral. The transformation, rather than resulting in a series of discrete lines, yields a continuum of amplitudes versus frequency. The form of the Fourier Integral is similar to that of the series with the summation going to an integration. Customarily, the exponential form is used. Thus, the phenomenon is described by

$$y(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} Y(j\omega) e^{j\omega t} \, d\omega \tag{2-9}$$

where the Fourier transform $Y(j\omega)$ is in general complex and is given by

$$Y(j\omega) = \int_{-\infty}^{\infty} y(t) e^{-j\omega t} \, dt \tag{2-10}$$

The complex transform $Y(j\omega)$ can be expressed as a real modulus and exponential argument. Thus,

$$Y(j\omega) = \left| Y(j\omega) \right| e^{j\theta(\omega)} \tag{2-11}$$

where $\left|Y(j\omega)\right|$ is the amplitude spectrum of the transformation and $\theta(\omega)$ carries the phase information.

Although one cannot speak of the energy of a specific harmonic when dealing with the transient phenomenon, one can speak of an amplitude or energy density as a function of frequency. Thus, one can identify regions of energy concentration, rates of rolloff, etc. From this information, system bandpass requirements for optimum signal-to-noise ratios can be established.

D. Signal Characteristics in the Frequency Domain

1. Ideal Signals

Signals which occur in an ideal world fit into seven basic categories. All signals are made up of combinations of these seven basic types. The seven are: stationary periodic, stationary complex, quasi-sinusoidal, stationary random, transient,

non-stationary periodic or complex, and non-stationary random.

a. Stationary Periodic

The simplest signal is described by

$$F(t) = F(t + T) \tag{2-12}$$

where T is a constant. The signal thus repeats identically whenever $t = t + T$. The smallest value of T for which Eq. (2-12) holds is called the primitive period, or period, T_o. The fundamental frequency of a periodic wave, f_o, is equal to $1/T_o$. A common example of a periodic signal is a sine wave.

$$F(t) = K \sin(\omega t + \phi) \tag{2-13}$$

where

K is a constant peak amplitude

ω is a constant angular frequency, equal to $2\pi f_o$

ϕ is a constant phase angle

Any stationary periodic signal of finite amplitude with but a finite number of discontinuities in one period may be expanded in a Fourier series.

b. Stationary Complex

A stationary complex signal is one which is produced by the addition of a combination of incommensurable periodic signals. An example is

$$F(t) = \cos \omega_1 t + \cos \omega_2 t \tag{2-14}$$

which is not periodic if ω_1 and ω_2 are incommensurable (for example, $\omega_1 = 1$ and $\omega_2 = \sqrt{2}$). The particular function shown is almost periodic, which means that the difference between it and an approximating periodic signal can be made arbitrarily small, during any finite time of interest, if the period of the approximating signal is made sufficiently long.

c. Quasi-Sinusoidal (Phase-Coherent Function)

A quasi-sinusoidal signal is a sinewave-like signal whose amplitude, frequency and/or phase are slowly varying func-

tions of time. Such a function may be periodic or almost periodic. An example is an amplitude- or frequency-modulated signal.

d. Stationary Random

The previously discussed signals may be determined for all time by a knowledge of the values and behavior of a relatively small number of coefficients. This is not true of a random signal whose amplitude at any time is a random function. The characteristics of a random signal thus must be defined in terms of statistics and statistical averages. A stationary random signal is one whose statistical functions are constant with time. A given stationary random signal is describable in the amplitude domain by its probability density function and in the frequency domain by its power-spectral density function.

e. Transient

These signals are time varying functions whose significant behavior occurs over a restricted time interval.

f. Non-Stationary Periodic or Complex

These signals are similar to stationary periodic or stationary complex; however, the parameters of the signal (amplitude, frequency, phase of each component) are functions of independent variables, which may be random.

g. Non-Stationary Random

A non-stationary random signal is one whose statistical functions vary with time. The most common physical situations encountered in noise-control engineering are of this type.

2. Real Signals

Any real signal, if examined in sufficiently fine detail, is

a. Transient, since it must have begun at some time

b. Non-stationary random

The essence of signal analysis is to determine the category to which a particular signal can be fitted and the parameters which describe its behavior with sufficient accuracy for the in-

tended purpose of the analysis. Thus, a signal from a crystal-controlled oscillator would be considered to be stationary periodic if used as a local oscillator for a transmitter, but quasi-sinusoidal, or even non-stationary periodic, if used as a secondary frequency standard.

This is a very real problem. Consider the analysis of the sounds of a truck moving down a road apparently at constant speed past a fixed microphone. The truck engine is turning at 30 rps, and being a six-cylinder, four-cycle engine, some cylinder fires every 1/90 of a second.

The truck radiates acoustical energy from the engine, exhaust and body, excited by the seemingly periodic firings of the engine, the random irregularities of the road surface and the periodic irregularities of the tires and drive train. But the body vibrations might be quite regular, as determined by the suspension system. Engine firings are frequently not very periodic, since the apparent 90 Hz fundamental output of the engine is actually a summation of six non-stationary periodic signals of 15 Hz, and responses can be measured in the vicinity of many multiples of 15 Hz. The amplitude of the entire sound ensemble is a function of both the changing distance to the microphone and the changing aspect. Furthermore, the truck signals will be intermixed with such other signals as the background sound environment, electronic system noise, etc., and will be affected by the Doppler effect.

The analysis of the sounds of a real object therefore is not a simple matter. It is often necessary to analyze the same data many times in order to choose different sets of measurement parameters to bring out different characteristics of the signal. For example, in the case of a truck, spectrum analysis with 1 Hz resolution would accurately locate the line frequency structure, while measurement with 4 Hz resolution would more accurately determine line amplitude structure and variation. Use of a tracking filter or high-speed plotter may permit determination of other variations in structure. For parameters which are random functions, averaging is necessary to determine mean parameters and to obtain some measure of statistical reliability; however, the signal may be such that desired averaging time is not available. (If the truck goes 15 mph and the road is 220 feet long, then only 10 seconds of data are available, for example.)

It is evident that the type of analysis used for real signals requires considerable judgment and experience on the part of the analyst.

3. Impulsive Signals

Impulsive signals are a class of transient signals which are characterized by relatively short duration and which are non-repetitive within the time-span of interest. Such signals may be analyzed in the frequency domain using the Fourier Integral Transform as defined in Eq. (2-10).

The Transform is a continuum of complex amplitudes based on the nature of the entire impulse, over all time. Therefore, it does not show how the amplitude characteristics develop as the impulse takes place. It seems sensible and useful to associate a spectrum with every instant of time, as the impulse develops. Such spectra present conceptual and theoretical difficulties. The difficulties basically revolve about the question, "At time instant, τ, what slice of impulse is to be spectrum analyzed?" One answer to this question is to reduce the slice of impulse that exists from time $-\infty$ to time τ.[6] For this type of analysis, a Running Fourier Transform is used.

$$G(j\omega) = \int_{-\infty}^{\tau} g(t)\, e^{(-j\omega t)}\, dt \qquad (2\text{-}15)$$

where

$G(j\omega)$ = the running transform

$g(t)$ = the amplitude-time waveform of the impulse

ω = angular frequency, $2\pi f$

f = frequency

t = time

τ = running variable

A feature of this choice of instantaneous spectrum is that it approaches the true Fourier Transform of the entire waveform, in the limit, as t approaches ∞.

[6] C. H. Page, "Instantaneous Power Spectra," J. Appl., Phys., Vol. 23, No. 1, January 1952, pp. 103-106

The spectra resulting from Eq. (2-15) may be of two general types:

1. Spectra containing a zero frequency component

2. Spectra not containing zero frequency component

In order for the spectrum of a time waveform to have a zero frequency component, the waveform must have some residual area when positive and negative areas under the entire impulse are added [7] Such amplitude-time waveforms are not theoretically possible in free air since free air is not a medium that can transmit a pressure-time waveform like a unit step function. This means that free air will not transmit sound-pressure changes of zero frequency. Theoretically, then, the medium can only produce impulses that have no DC frequency component.[8]

4. Practical Considerations of Impulse Signal Handling

In practice, the limited dynamic ranges of the recording and processing equipment can result in a distorted impulse being finally reduced. The distorted impulse may have a net area under the waveform, and therefore a DC component in its spectrum. For minimum distortion, FM recording [9] and equipment having a large dynamic range and low-frequency cutoffs are necessary.[10]

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[7]F. Skodde, "Low Frequency Measurements Using Capacitive Transducers," B & K Technical Review, No. 1, 1969, p. 14.

[8]H. P. Olsen, "Frequency Analysis of Single Pulses," B & K Technical Review, No. 3, 1969, p. 8; see also, W. B. Snow, "Survey of Acoustic Characteristics of Bullet Shade Waves," IEEE Trans. on Audio & Electroacoustics, Vol. AU-15, No. 4, 1967, p. 163.

[9]E. D. Sunde, "Theoretical Fundamentals of Pulse Transmission - I," Bell System Technical Journal, May 1954, p. 773 ff; see also, K. G. Kittleson, "Introduction to Measurement and Description of Shock," B & K Technical Review, No. 3, 1966.

[10]M. J. Crocker and L. C. Southerland, "The Effects Upon Shock Measurements of Limited Response Instrumentation," Wyle Laboratories, Research Staff Report WR 65-1, January 1965.

2.2.D-2.2.E

The low frequency cutoff should be

$$f_{\ell} \leq \frac{1}{40\pi\, t_o} \qquad (2\text{-}16)$$

where

f_{ℓ} = the lower frequency cutoff (Hz)

t_o = the positive phase duration (Sec)

This provides less than a 4% degradation in the duration of a primarily unipolar type of impulse.

The high frequency cutoff preferably should be[11]

$$f_h \geq \frac{125}{\pi\, t_o} \qquad (2\text{-}17)$$

where

f_h = high frequency cutoff (Hz)

This provides less than a 4% degradation in the peak amplitude of the unipolar type of impulse; albeit a more modest requirement, $f_h = 10/t_o$ often is adequate.[8]

E. Practical Approach to Frequency Analysis

1. Limitations of Laboratory Analysis

Just as real world signals become available for analysis through the use of recorders which have limited frequency-domain and time-domain capabilities, so also do the practical limitations of laboratory instruments affect the results of frequency analysis. It is important that the analyst under-

[11] See footnote 10, p. 2-19

stand these limitations. In general, the limitations come from (a) the limitation of sample length of data chosen ("truncation") which affects the statistical reliability of the result, (b) from the bandwidth of the analysis equipment, and (c) from the number of passes of the data sample used in the analysis. The effects of analysis bandwidth will be discussed in more detail in Section 2.3. In this section, the effect of data truncation, signal stability and the statistical reliability of the results are discussed.

2. The Effects of Repetitive Scan

Every sample of signal which is analyzed is necessarily of finite duration. If the sample is of length T, then the effective resolution cannot be finer than 1/T Hz. When sweeping type analyzers are used (see Section 2.3), the sample is repeated many times while the filter of the analyzing equipment slowly sweeps through the frequency range of interest. Thus, what was in reality a transient section of data, because of the type of equipment used, becomes a repetitive signal. Any Fourier analyzer which does not weight the data (see following subsection, "The Effects of Data Truncation") will treat the signal as if it were one cycle of a periodic disturbance having a period of T seconds. (Most fractional octave sweeping filters and wave analyzers are of this type.) The result is a Fourier Series based upon a fundamental frequency of 1/T Hz. For this reason, the resolution bandwidth of the analysis cannot be less than the spacing between harmonic multiples of the series, i.e., 1/T Hz.

For random data, if the repetition rate is once per second, the fundamental frequency of the series is 1 Hz and the analysis with 1 Hz wide band filter yields spectrum level directly.

3. Fitting the Data to the Range of the Analyzer

The frequency content of data may be multiplied by a factor K to fit the analyzer range, by running the tape at K-times the speed at which the data was recorded. The real frequency is obtained by dividing the analyzer readings by K. The band-

width of analysis should also be divided by K.

4. The Effects of Data Truncation

Since only a limited sample of data is taken for analysis, there is an unavoidable analysis error. This is true even if the signal being analyzed is stationary periodic. The cause of error can be understood by realizing that a T-second sample will appear to any analyzer as a periodic signal with period T multiplied by a window function F(t), where

$$F(t) = 1,\ 0 \leq t \leq T \quad \text{and} \tag{2-18}$$

$$F(t) = 0,\ \text{otherwise} \tag{2-19}$$

Consequently, the Fourier Transform produced by an ideal analyzer of the T-second sample will be that of the equivalent Fourier Series of the sample convolved with the transform of the function F(t). If the original signal is a sine wave of frequency f_o, then the Fourier Transform $Y(j\omega)$ of a T-second portion of the sine wave is

$$Y(j\omega) = \frac{\sin \pi(f - f_o)T}{\pi(f - f_o)T} \tag{2-20}$$

This has pronounced sidelobe structure (Figure 2-1) at intervals of 1/T Hz. The first sidelobe is down only 13.2 dB and the 320th sidelobe is down 60 dB.

Since the desired form of transform of the original sine wave is simply a single line at f_o, it is customary to "shape" the window function, F(t), so that the transform of a finite sample of a sine wave closely resembles a single line.

That is, the window function is made other than rectangular in order to minimize its effect on the analysis. This is done by weighting the data samples nonuniformly from $-T/2 \leq t \leq T/2$. The effect is to widen the resolution bandwidth of the analyzer somewhat from 1/T Hz, but in return to drastically reduce the sidelobe structure. Modern real time analyzers frequently employ this weighting technique (see subsection 2.3.D).

Three common windows used in analog spectrum analyzers are

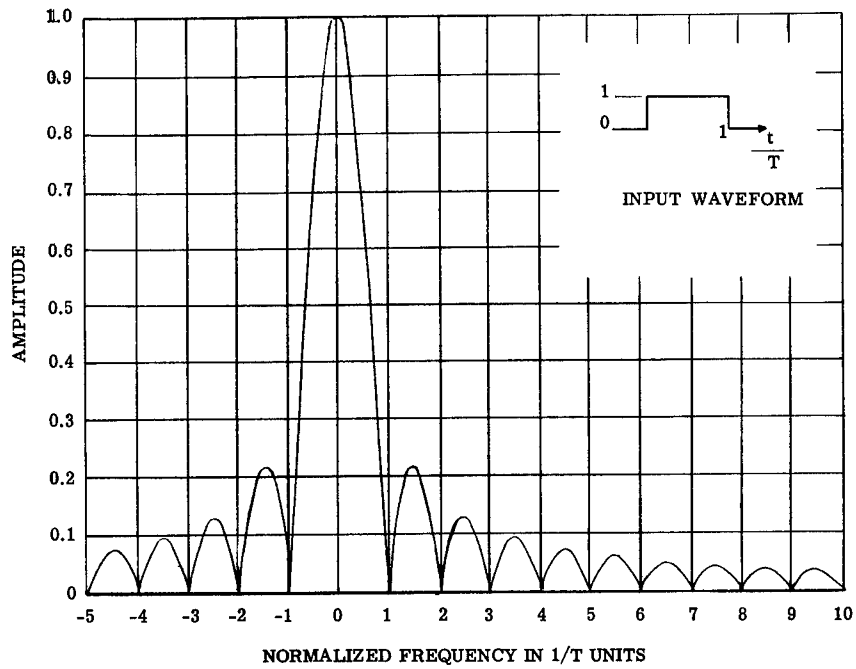

Figure 2-1. Spectrum of Unweighted Window

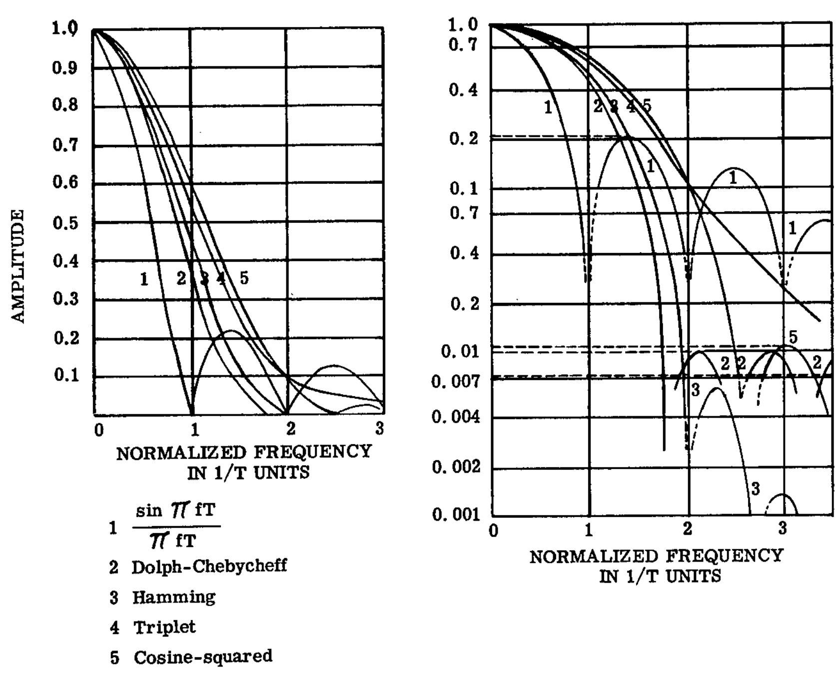

Figure 2-2. Comparison of Various Weighting Functions

2.2.E

the Triplet having a form

$$F(t) = e^{-(\pi/T)t} \cos^2 \frac{\pi t}{T} \qquad -\frac{T}{2} \leqq t \leqq \frac{T}{2} \tag{2-21}$$

the Hamming

$$F(t) = 0.54 + 0.46 \cos \frac{2\pi t}{T} \qquad -\frac{T}{2} \leqq t \leqq \frac{T}{2} \tag{2-22}$$

and the Cosine-Squared

$$F(t) = \cos^2 \frac{\pi t}{T} \qquad -\frac{T}{2} \leqq t \leqq \frac{T}{2} \tag{2-23}$$

The Triplet response has no sidelobes but a broad main lobe (1.73 times that of an unweighted window at the 3 dB down point, 15/T at -40 dB and 50/T at -60 dB).

The Hamming has a slightly narrower main lobe (1.5 times that of an unweighted window) and a strongest sidelobe (the third) level of -42.8 dB, with other sidelobes falling off as 1/f from the third.

The Cosine-Squared window has an intermediate width main lobe (1.59 times that of an unweighted window) and a strongest sidelobe (first) level of -31.4 db, with other sidelobes falling off as $1/f^3$ from the first, so that the fourth sidelobe is down 54 dB. For comparison, the fourth sidelobe of the Hamming is down 43 dB. Figure 2-2 shows a comparison of several weighting schemes.[12]

5. Effect of Signal Stability[13]

Frequently, as with data stemming from moving sources, the real analysis signal contains strong line components whose frequency varies with time. It is important to understand how the analysis equipment treats these signals and the errors which occur thereby.

The effect of the frequency stability of the source can be seen by examining the analyzer response to a sliding tone. A sliding tone is defined as a quasi-sinusoid whose frequency

[12]C. L. Temes, "Sidelobe Suppression in a Range-Channel Pulse Compression Radar," IRE Trans. on Military Electronics, April 1962, pp. 162-169.

[13]W. Gersch and J. M. Kennedy, "Spectral Measurements of Sliding Tones," IRE Trans. CT-7, August 1960.

changes linearly with time.

$$s(t) = \exp 2\pi j(f_o t + kt^2) \qquad -\frac{T}{2} \leq t \leq \frac{T}{2} \qquad (2\text{-}24)$$

The total amount of frequency slide (Δf) is kT where k is the rate of frequency sweep in Hz per second, in the analysis time interval T, so that the frequency excursion of the signal in resolution bandwidths, 1/T, is

$$n = kT^2 \qquad (2\text{-}25)$$

The spectrum for a given value of n depends both on n and on the window function F(t). The spectrum of a sliding tone with a rectangular window function is shown in Figure 2-3. For small n, the major effect is a sizable increase in sidelobe level and a consequent broadening of the spectrum line, especially in a logarithmic plot. For large n, say $n > 50$, there is an approximately equal distribution of energy across the band. Thus, the spectrum of a sliding tone having n = 100 will be approximately 100 resolution bandwidths wide and will have a spectrum level of 20 dB down from that of an equal energy stable signal.

For intermediate values of n, the Fourier Transform is quite complex (see Figure 2-4, which shows responses for a one-sided slide). There is a reduction in the peak amplitude of the spectrum (relative to that of a steady tone) according to the following table:

n	Amplitude Reduction, dB
0	0
1	0.3
2	1.0
4	4.0
6	7.7
8	9.6

This is less than would be expected by equal power division over the frequency range. The sidelobe structure broadens and, for n = 6 and higher, there may be a minimum at the center frequency rather than a maximum.

For Cosine-Squared weighting (see Figure 2-5) the effect is far less, both in terms of amplitude response and in terms of sidelobe structure. A moderate slide actually smooths over the sidelobes. For n = 4, there is a peak loss of about 1 dB, and

almost nothing for n = 2. The Triplet shows similar behavior, with responses down as follows:

n	Amplitude Loss, dB
0	0
1	0
2	0.2
4	0.7
8	3.0
16	4.8
32	8.0

Therefore, for Triplet or Cosine-Squared weighting, a total frequency instability of 4 resolution bandwidths per analysis time will produce relatively unimportant effects on broadening and amplitude reduction.

When analyzing a signal of varying frequency, it is necessary to widen the resolution bandwidth and reduce the analysis time to achieve an accurate indication of the peak spectral line level. For example, suppose one is analyzing the signal from an accelerating vehicle and the engine speed is changing 10% per second, then a 30 Hz nominal line would change 3 Hz/second. If an analysis bandwidth of 1 Hz were used and the time sample were 1 second, n would be 3 which for Triplet weighting would result in a line level accurate to about 0.5 dB. For the fifth harmonic, however, the frequency slide would be 15 Hz/second and the resolution and sample time would result in n = 15, resulting in an error of almost 5 dB. To get an accurate indication of the level of the fifth harmonic, one should increase the resolution bandwidth to 2 Hz and reduce the sample time to 1/2 second. This results in n equal to 3.75 and reduces the error to about 0.6 dB.

6. Statistical Reliability

The emanations from many sources and backgrounds are basically random in nature. The spectra from such sources will show statistical fluctuations, and the standard deviation of these fluctuations depends on the number of degrees of freedom of the analysis. The number of degrees of freedom of a single Fourier analysis of duration T and resolution 1/T is two, since for each spectral component an amplitude and a phase may be independently specified. The number of degrees of freedom may be increased by averaging or integrating spectra of a number of independent time samples of the signal. Since the time taken to analyze one sample of data is equal to 1/B where B is the anal-

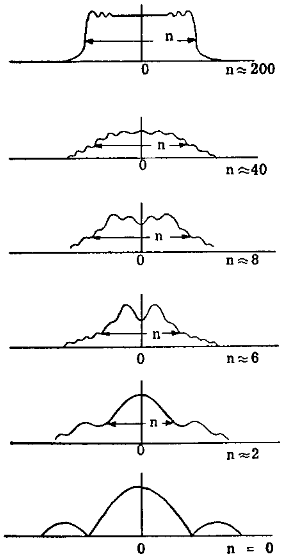

Figure 2-3. Spectrum of a Sliding Tone, Uniform Weighting

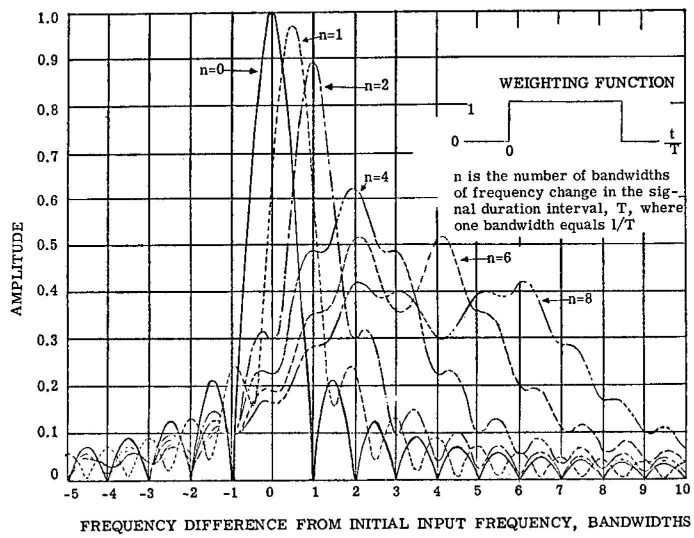

Figure 2-4. Spectrum of a Linear One-Sided Frequency Slide, Uniform Weighting

$$I(h,n,b) = \left| \int_{-\frac{1}{2}}^{\frac{1}{2}} \cos^2 \pi x \exp j\pi (nx^2 - 2bx)dx \right|$$

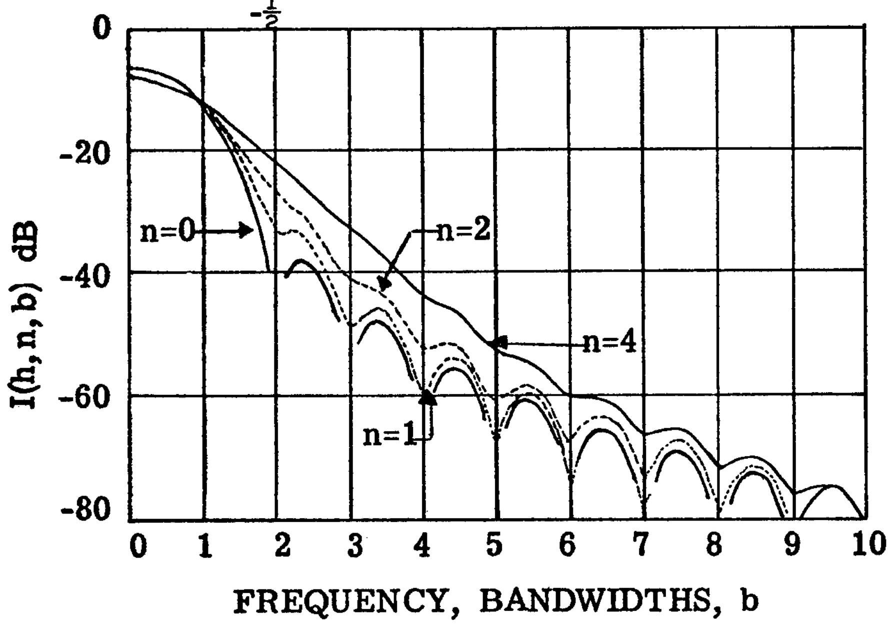

Figure 2-5. Spectral Response of the Filter $h(x) = \cos^2 \pi x$ to a Sliding Tone

ysis bandwidth in Hertz, in a total integration time T, BT independent analyses can be made. Thus, the number of degrees of freedom is approximately twice the bandwidth-integration time product of the analysis, that is

$$k = 2BT \tag{2-26}$$

The number of degrees of freedom, k, is equal to 2 BT for "flat" spectra such as from random noise. When a line spectrum is evidenced, the number of degrees of freedom is reduced. This is apparent in the extreme case of a single sinusoid when there are only two degrees of freedom (amplitude and phase of the one spectral line) regardless of integration time or bandwidth. When the spectrum of a signal contains both line and random components, the number of degrees of freedom of the ensemble is[14]

$$k = \frac{\left(\sum^{n} b_i\right)^2}{\sum^{n} {b_i}^2} \tag{2-27}$$

where

k is the number of degrees of freedom

b_i is the value of the i^{th} component (either amplitude or phase)

n is the number of data points, i.e., total number of amplitude points and phase points for total integrator time

For a given n, k maximizes when all b_i are equal and then

$$k = \frac{n^2 {b_i}^2}{n\, {b_i}^2} = n = 2BT \tag{2-28}$$

For this reason, it is desirable to "pre-whiten" the data before analysis to improve the statistics. This is accomplished by filtering out the strong line components and shaping the frequency response of an equalizer to flatten the spectrum.

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[14]Yale J. Lubkin, "Lost in the Forest of Noise," Sound and Vibration, November 1968, pp. 23-25.

After analysis, the complementary equalization is applied to the spectrum to correct it. The value of the coherent lines is established by separate analysis and reinserted into the spectrum. When the lines are not removed from the spectrum, the value of k in the vicinity of the line is decreased by a factor approximating the ratio of the line width to the analysis bandwidth.

For random data, the anticipated spread in the expected values of spectral components is called the confidence interval and is a function of the probability of the spectra of the random data exceeding this spread. It can be shown that the standard deviation of the value of the spectral lines from Gaussian data is given by

$$\sigma = \frac{1}{\sqrt{BT}} \tag{2-29}$$

Based on this formula, the following table of confidence intervals is calculated. Note that this uncertainty applies strictly only to Gaussian random spectra and is due to the nature of random signal and not to the analysis.

CONFIDENCE INTERVAL IN dB

		Confidence Level							
		80%		90%		95%		99%	
BT	k	+	-	+	-	+	-	+	-
1	2	3.6	9.8	4.8	13	5.7	16	7.2	23
2	4	2.9	5.8	3.6	7.5	4.5	9.2	5.7	13
2.5	5	2.7	4.9	3.5	6.4	4.1	7.8	5.3	11
4	8	2.2	3.6	2.9	4.7	3.4	5.7	4.4	7.8
5	10	2.0	3.1	2.6	4.1	3.1	4.9	4.0	6.7
10	20	1.5	2.1	2.0	2.7	2.3	3.2	3.0	4.3
25	50	1.0	1.3	1.3	1.6	1.6	1.9	2.0	2.5
50	100	0.7	0.8	0.9	1.0	1.0	1.2	1.3	1.6
100	200	0.5	0.5	0.7	0.7	0.8	0.8	1.0	1.0
250	500	0.4	0.4	0.4	0.4	0.5	0.5	0.7	0.7
500	1000	0.3	0.3	0.3	0.3	0.4	0.4	0.5	0.5

For example, for any particular resolution element (spectral line) for k = 8, there is an 80% probability that the measured value will lie between +2.2 and -3.6 dB of the true value.

It is desirable, then, to integrate the signal spectrum for as long a time as possible in order to increase k for a given resolution bandwidth and so decrease the confidence interval. If the signal is not stable, then the length of time that one can integrate is limited, and the confidence interval is correspondingly wide. Alternately, one may increase the resolution bandwidth to raise k when signal instability prevents increasing the integration time. In this case, one trades off frequency resolution for amplitude accuracy in the band.[15]

F. <u>Signal Characteristics in the Time Domain</u>

1. <u>General</u>

The importance of time domain analysis lies in its ability to aid directly in the design of electronic detection devices. Often, it is insufficient to know the power spectrum density of the signals to be detected because there may exist noise sources with significant energy in the same portion of the spectrum as the desired signal. The key difference between the desired signal and the noise may reside in the relative phases of the components. For example, both white noise and a delta function have a flat power density spectrum, i.e., equal power per unit frequency (see Figure 2-6). (Strictly speaking, the frequency transform of delta function extends from $-\infty$ to $+\infty$.) There is, however, a coherent relationship among the phases of the components which constitute a delta function and a random relationship between those of white noise. The result in the time domain is a random distribution of levels for white noise and, for the delta function, a zero level everywhere except at zero reference time where the level approaches infinity. These two signals can be distinguished readily in the time domain by their different amplitudes but, because of equivalent power density spectra, it is difficult to distinguish them in the frequency domain.

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[15]This is a manifestation of Uncertainty Principle as applied to signal analysis. See B. B. Bauer, "Octave-Band Spectral Distribution of Recorded Music," J. Audio Eng. Soc., April 1970.

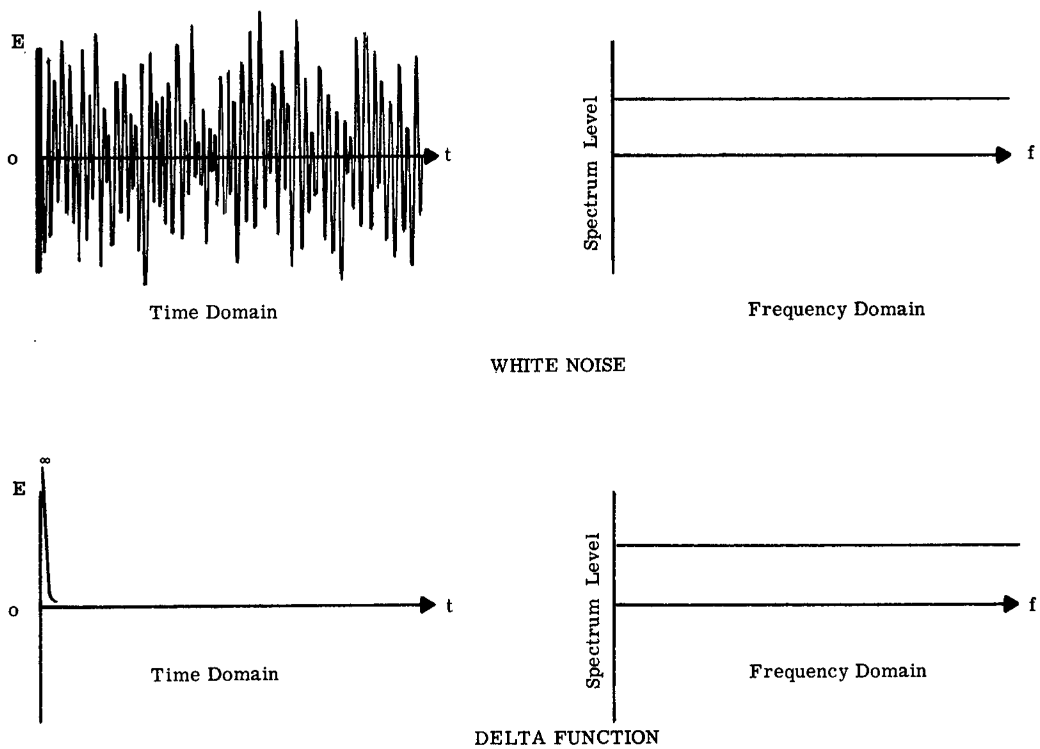

Figure 2-6. Spectral Comparison of White Noise and Delta Function

Sometimes, a key difference between two signals is a difference of power level variation with time. Thus, an accelerating truck exhibits a variation in power level as it accelerates in turn through each gear with sudden decreases in level as the gears are shifted. Such variations can be used to distinguish the truck from other sources. The variation in power level can be measured on a broadband basis, but it is frequently best to restrict the band to that frequency region where the most distinguishable variation occurs. This ability to detect those sources which change

with time in some prescribed manner provides a powerful identification tool.

The time-domain information discussed above is from a single sensor. A very powerful application of time-domain analysis is correlation in which the product of two signals is integrated with time to determine the degree to which they correspond in time. The two signals may be received by two different spaced-apart sensors (spatial correlation) or from the same sensor with one of the signals delayed in time with respect to the other (time correlation). The result of correlation is an improvement in signal-to-noise ratio since, if the experiment is properly designed, the correlated signals from the same source will produce a greater output than correlated random signals.

A detection scheme which uses multiple sensors operating in different signals fields (i.e., acoustic, seismic, infrared, etc.) can provide much improved signal identification, without requiring the use of sophisticated correlation systems. In these cases, it might be considered that a detection coincidence criteria for the individual sensors constitutes a simple form of correlation. In any case, coincidence detection in multiple signal fields within the scope of time-domain analysis is based on the interrelationship of the variables with time and not on their individual characteristics.

2. Time-Domain Characteristics

It is helpful to define certain terms used in time-domain analysis especially since they are frequently used with different meanings in different texts.

Envelope. The envelope is the locus of points which are the extremes of a function between axis crossings. Typically, the envelope is the pair of curves one of which joins each of the peak positive signal excursions and the other of which joins each of the peak negative signal excursions (Figure 2-7).

Rectified Envelope. Frequently, the function of interest is the envelope of the signal after rectification. In such case, the type of rectification should be designated, e.g., positively rectified envelope, negatively rectified envelope, or full-wave rectified envelope. The rectified envelope of a signal will depend upon the amount of "smoothing," if any, provided by the low pass filter after rectification.

Rise Time. This is the time required for the specified time function or envelope to increase from 10% to 90% of its maximum value. An accurate estimate of this parameter requires a good signal-to-noise ratio. When the nature of the signal is such that no such idealistic approximation can be made, the definition of the 10% value for the signal is often specified in whatever form is most useful to the circuit designer. The more common definitions are the point at which the rms signal exceeds the rms noise by 10% or when the peak signal exceeds the peak noise by 10%. In such cases, the conditions under which the determination was made must be stated (Figure 2-7).

Fall Time. This is the time required for the signal to diminish from 90% of its maximum amplitude to 10% of its maximum. The same signal-to-noise limitations discussed concerning rise time are imposed on any attempt to precisely measure the fall time parameter (Figure 2-7).

Signal Duration. The signal duration is considered to be the time period between the 10% point of the rise characteristic and the 10% point of the fall characteristic. The usefulness of this parameter is associated with the design of AGC networks. It can aid the designer in deciding the amount of signal suppression to be expected from a particular circuit (Figure 2-7).

Signal Gradient. In those instances where the signal cannot be approximated by a set of straight lines representing "rise," "duration," and "fall," it is necessary to describe the variable structure in a somewhat more comprehensive manner. One such description can be obtained by sampling the signal envelope and plotting the amount of change between adjacent samples. This procedure produces a plot which gives the designer a complete description of the manner in which the signal level has fluctuated with time. There are three areas of decision which must accompany any analysis of this type. These are: the detector smoothing, the sampling rate, and the necessity of smoothing through the averaging of several adjacent samples.

Effective Bandwidth. The electronic restrictions placed upon the frequency components which are examined for their time characteristics establish the effective bandwidth of the system. The bandwidth chosen may not coincide with the spectral peaks found by frequency analysis since the objective is to find an area of the spectrum in which time fluctuations in the signal are in some way unique.

G. Correlation in the Time Domain

An important analytical technique for processing data is correlation. Similar to the frequency-domain Fourier transformation mentioned in subsection 2.2.C, correlation involves an integration of the input signal multiplied by a companion signal. Unlike the Fourier transformation which used trigonometric functions as a companion multiplying signal with the integrated result becoming a function in the frequency domain, for correlation the companion signal can be a variety of functions including the input signal itself. The result is now a function of lag time between the input and the companion signal. This can be expressed mathematically as

$$G(\tau) = \int_{-\infty}^{\infty} g(t + \tau)\, f(t)dt \qquad (2\text{-}30)$$

where $g(t)$, $f(t)$ are the input signal and the comparison signal, respectively, and τ a time displacement between the functions for which the correlation is computed.

One of the most important uses of correlation is for signal-to-noise ratio enhancement.[16] It is especially effective for repetitive signals. If $g(t)$ consists of two parts, noise $n(t)$ and signal $s(t)$, and $f(t)$ is of the same characteristics as $s(t)$, then the improvement in signal-to-noise ratio of $g(t)$ is approximately

$$R = 10 \log_{10} \frac{N}{1 + 4\rho_i^{\,2} + 2\rho_i^{\,4}} \ (\text{dB}) \qquad (2\text{-}31)$$

[16] Y. W. Lee, "Application of Statistical Methods of Communication Problem," T. R. No. 181, Research Laboratory of Electronic Technology, M.I.T.

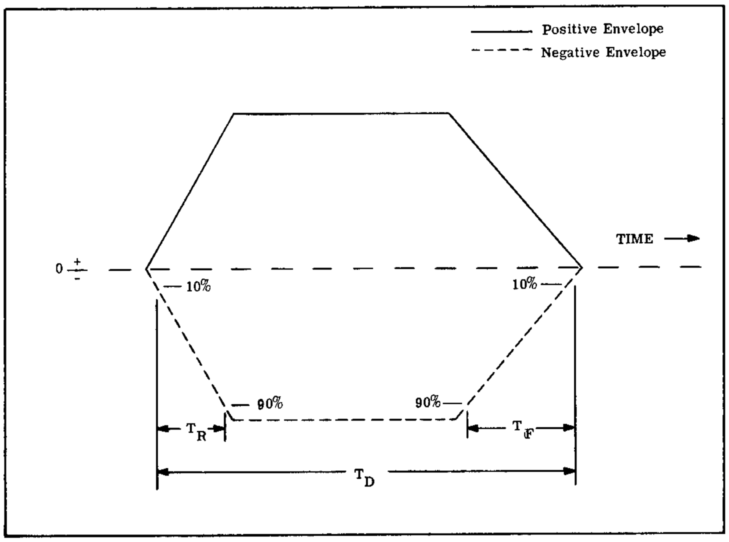

Figure 2-7. Pictorial Representation of Signal Envelope Showing Rise Time (T_R), Duration (T_D), and Fall Time (T_F)

where N is the number of sample products which the correlation performs and ρ_i is the rms noise-to-signal ratio. Similarly, if g(t) and f(t) are identical signals (auto correlation), the expression becomes

$$R = 10 \log_{10} \frac{N}{1 + 2\rho_i^2} \text{ (dB)} \qquad (2\text{-}32)$$

The net result often provides a signal enhancement of 10 or more dB allowing a more useful signal to be analyzed. The time-domain characteristics of the signal are modified by the correlation, however. A simple example of this modification would be the case of a repetitive square wave buried in noise which, after correlation, would be a repetitive triangular wave. Although the signal enhancement property alone is of value in data reduction and analysis technology, the ability of correlation to determine lag times for a signal which is transmitted to different sensors through a medium can also be of analytical value.

This is discussed in more detail in subsection 2.6.D. Finally, the presence or lack of correlation of the signal with itself can give guidance as to the similarity of the two propagation paths, a technique similar to that used in circuit analysis to determine the transfer function of an unknown circuit.

H. Utilization of Time Domain Information

1. Automatic Gain Control

In order to produce a sensing system which can accommodate wide variations in both signal level and masking noise sources, it is necessary to apply some means of signal level control. In the majority of these systems, this is accomplished by integrating a portion of the energy from the sensor and applying it to a negative gain-controlling feedback network. The place in the electronic circuit from which this control voltage is derived is chosen by the designer from the knowledge of the characteristics of the source to be detected. It is not unusual for control signals to be derived from several areas of the circuit so as to provide maximum noise rejection without degrading the detection reliability or increasing the false alarm rate. The areas most commonly examined for such control voltage are:

a. spectral energy in separate noise sensing channel

b. the integrated average energy in the data channel having time constants longer or shorter than the desired signal, and,

c. a filtered post-detection channel for sensing signals not having the same envelope spectrum as the desired signal.

The last technique may actually be a differentiating network which can be considered as a post-detection high-pass filter. A knowledge of the signal rise time characteristic will allow the designer to optimize a differentiation network for a particular class of signals. This same information is necessary to design the attack times of AGC networks to be applied to the signal channel.

If the entire signal envelope has to be retained to achieve reliable detection and identification of the signal, it will be necessary to examine closely the duration of the signal to assure that an AGC network is applied which acts as rapidly as possible without degrading the desired signal. The rate at which the AGC network recovers after the passage of a signal can be optimized if the fall time of the signal is known. In those instances where the manner in which a signal decays is unique enough to aid the design of identification logic, this consideration is especially important. It may be necessary to follow the controlled amplifier by a differentiator of some sort which has been optimized

for the desired negative-going envelope (toward zero). Such considerations are also vital if the sensor is required to detect many successive targets, since an improperly chosen decay time may desensitize the system for the later signals of the group.

2. Post-Detection Filtering

If the signal of interest has an envelope shape which itself contains reasonably unique spectral characteristics, it is often beneficial to apply some form of post-detection filtering. The reason for mentioning this subject under the topic of time domain is twofold: (a) the signal being filtered is the time envelope of the signal--not the original signal itself, and (b) the application of differentiation networks (special high-pass filter) is directly related to the time structure of the signal.

It is entirely possible that a separate time-domain analysis may be required on the output of the post-detection filtering network. An example of this sort is a detector for a signal whose envelope fluctuates at a rate which is a direct function of the source location. . Here, it is necessary to design a post-detection combination filter whose individual AGC rates are determined by the sweep rate of the pulsating signal. This example is mentioned to indicate the areas where time-domain studies much simplify design procedures. The application of differentiating networks for the examination of decay characteristics is often overlooked. The existence of such unique signals is not nearly as rare as one might think.

2.3 FREQUENCY DOMAIN DATA ANALYSIS

A. Introduction

1. Types of Analysis

A variety of devices has been developed for measuring power spectra of a stationary periodic signal[17]; however, all analog analyzers employ the same basic functions--filtering, detection, averaging, and displaying the result on a meter or other output device. In some cases, the signal is filtered by a narrow-band filter, for which the center frequency is varied to obtain the

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[17] If the signal is not stationary periodic, a portion thereof is stored and repeated so as to present a stationary periodic signal to the analyzer.

2.3.A

power variation with frequency. In other cases, a band of fixed filters is used, giving simultaneous outputs at a number of center frequencies. A third technique is to use a "heterodyne filter," in which the signal is heterodyned with a local variable-frequency oscillator and a narrow band of frequencies in one of the sidebands produced during modulation is filtered using a fixed-frequency filter. The effective analysis frequency is then controlled by changing the frequency of the variable-frequency heterodyning oscillator. The same device can be used for non-stationary signals. In this case, the results of analysis for any particular frequency vary with time, in a manner which depends on the output device used.

Usually, the first or second of the above-described methods are employed if percentage or fractional bandwidth characteristics are desired; the third method results in constant bandwidth analysis. The output of the filter, in all cases, is representative of those components of the signal near the effective center frequency of the filter (subject to the condition that the filter envelope response time is short compared to the signal duration.) If a heterodyne system is used, the filter output will differ from that of a non-heterodyne type only by a shift in center frequency; for instance, if the non-heterodyne output is a sinusoid

$$f(t) = A(t) \cos [\omega_o t + \theta(t)] \tag{2-33}$$

then the heterodyne output would be

$$f(t) = A(t) \cos [(\omega_c \pm \omega_o)t + \theta(t)] \tag{2-34}$$

where ω_c is the frequency of the heterodyning oscillator. The $\pm$ sign is chosen on the basis of whether an upper or lower sideband is used. The instantaneous power (short-term average) in either case is found from

$$p(t) = \overline{f^2(t)} = 1/2\ A^2(t) \tag{2-35}$$

The value of the power-spectrum density, as measured at the frequency ω_o, is then equal to the average value of $p(t)$. The averaging time now is chosen on the basis of the length of time during which the signal remains stationary.

In most cases, the envelope of the detected signal is not squared before averaging which results in the calculation of a voltage spectrum

$$E(f) = \overline{A(t)} \tag{2-36}$$

This procedure can be justified on the basis that the average and rms values of a narrow-band signal are related by a factor that remains relatively constant, regardless of the amplitude distribution of the signal before filtering. For instance, this factor for a sine wave is $2\sqrt{2}/\pi$ or 0.903. For a Gaussian signal (Rayleigh envelope) it is $\sqrt{\pi}/2$ or 0.887. The difference is less than 2%, even though the amplitude distribution for a sine wave is markedly different from a Gaussian distribution.

2. Presentation of Data

Selection of the proper information to consider (or display) is vital to proper interpretation of signal characteristics. Some of the types of displays which are in common use today are spectrum "snapshot," spectrum time plot, and single-line behavior.

a. Spectrum "snapshot." A spectrum is computed on the basis of an integration time, T, with selected analysis bandwidth. The display consists of an X-Y plot of spectral power density versus frequency. Examples of this display format are given in subsection 2.3.D (Figure 2-25). This approach gives a quantative understanding of the signal characteristics provided the spectrum is stationary, i.e., it does not change with time. Generally, an appropriate integration time is determined experimentally--if the results of the analysis do not change appreciably over one integration period, the choice is satisfactory.

b. Spectrum time plot. When the signal is not stationary, an intensity-modulated X-Y plot can be used to show the behavior of the spectrum as a function of time. Frequency is taken along one axis, time along the other, and intensity modulation is used to give the spectral level at a given frequency and time. The time-wise character of the signal under study gives insight into the physical processes of signal generation, provides an overview of the signal structure and its behavior, and provides a basis for rejection of signals from unwanted sources (since the behavior with time of signals from different sources will normally be different). The chief disadvantage of this type of display is that the information extracted from it tends to be of qualitative rather than quantitative nature. Examples of spectrum-time plots are given in subsection 2.4 (Figures 2-27 and 2-28). An alternate approach to the spectrum-time plot is the "3-D spectral display" which depicts the results of many sequential spectral analyses in a pseudo-three-dimensional manner. Such a representation is seen in Figure 2-29.

2.3.A

c. Single-line behavior. Devices, such as the phase-locked loop, can be used to study the behavior of a single "line." Typically, the phase of the data signal is compared against that of a signal produced by a local oscillator. The comparison voltage is used to adjust automatically the frequency of the local oscillator so that the two signals remain in phase. The comparison voltage is thus a measure of the instantaneous frequency of the data signal. Either the amplitude or the frequency of the "line" may be shown as a function of time in an X-Y plot. This type of presentation is valuable for studying the behavior of an individual "line," but gives little insight into the total signal characteristics.

3. Interpretation of Data

It is necessary to take a close look at the manner in which the signal analyses and noise analyses are performed to prevent serious misinterpretation of the results. If the signals and the noise are of the same character (for example, both are random in character), the results of their analyses by the same techniques will yield directly comparable results. Figure 2-8 shows such a case in which the signal and the noise spectra are continuous, and a direct overlay is useful as a guide to the system designer. A plot of the signal-to-noise ratio derived from Figure 2-8 is shown in Figure 2-9. By matching this curve with an input bandpass filter, the designer can begin to optimize the detectability of the signal.

a. Random Signals. In interpreting data, a problem may arise from the fact that the detected output usually is a function of the filter bandwidth. Practically, the spectrum levels are obtained with octave-, 1/3-octave-, 1/10-octave, or constant-bandwidth filters. The results of signal analysis obtained with different bandwidths are not directly comparable with each other. The resulting spectral plot is then said to be in terms of Spectrum Density Levels, often called simply "Spectrum Levels." It becomes necessary to modify the measured values by some proper factor so that the data taken with analyzers of different bandwidth can be compared meaningfully.

Bandwidth correction or "normalization" of random signals is accomplished by dividing the output power reading by the bandwidth. In terms of levels, this is equivalent to diminishing the level by 10 log Δf, dB, where Δf is the bandwidth of the filter in Hz. It should be noted that with constant fractional bandwidth filters, Δf increases with frequency f; thus $\Delta f = Ff$, where F

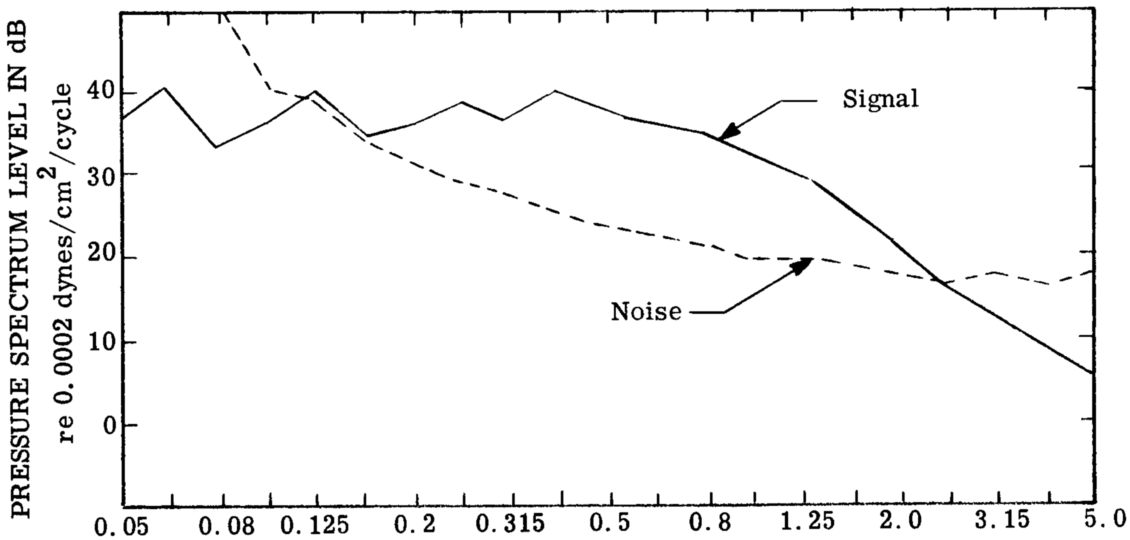


Figure 2-8. Pressure Spectrum Levels Obtained by One-Third Octave Analysis of a Continuous Signal and Masking Noise Source, Normalized to 1 Hz Bandwidth

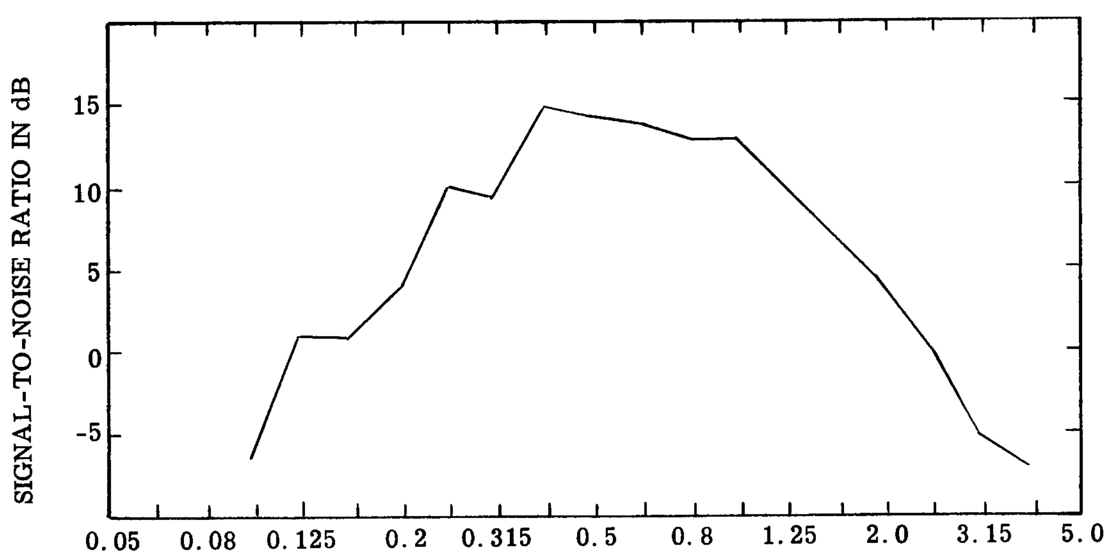


Figure 2-9. Signal-to-Noise Ratio Plot Derived from Superposition of Spectra Shown in Figure 2-8

2.3.A

is the fraction of bandwidth used, and the needed correction is -10 log Ff, to produce a result whose amplitude is equivalent to that produced by a filter with a 1 Hz bandwidth. For example, the analysis shown in Figure 2-10 was done with a 1/3-octave analyzer and a filter bandwidth correction was applied as described above. The effect of such a bandwidth normalization is shown in the dash line. Bandwidth normalization helps to prevent errors in interpretation. It should be pointed out that the uncorrected analyzer output gives the impression that the signal has a great deal of high frequency power when, in fact, the power density is very low in that part of the spectrum.

Strong single-frequency lines of a stationary spectrum are unaffected by the filter bandwidth, and the correction factor described in the preceding paragraph is not applicable to them, as discussed in subsequent paragraphs.

The process of bandwidth normalization is rooted in two basic concepts which are common to the field of "noise measurement." The first consideration is the characterization of noise in terms of its power-spectrum density. This form of presentation was defined by early researchers as the average power per-cycle bandwidth, thus requiring real-filter systems to be corrected if this form of presentation were desired. In addition, and closely related to this, researchers in the field of noise measurement needed a means of validating and comparing measurements made with different analyzing schemes. It was found that when "white" noise was applied to the real analyzers being used, and the individual filter output voltage was divided by the square root of the filter bandwidth (or dividing the power reading by the bandwidth), the resulting spectrum was indeed flat with frequency. This, then, provided a means of validating the many analyses schemes available, and increased the confidence in the physical interpretation of the data being presented by the discipline.

In some instances, this form of correction was designed into the analysis system by electrically de-emphasizing the data (3 dB/octave for 1/3-octave systems). In the data analyzed for the AWG effort, this concept is retained with all corrections being accomplished manually so that their magnitude could be changed to better suit the known physics of the data.

When the bandwidth, Δf, is given in terms of the upper and lower cutoff frequencies, the amount of correction, C, applied to the

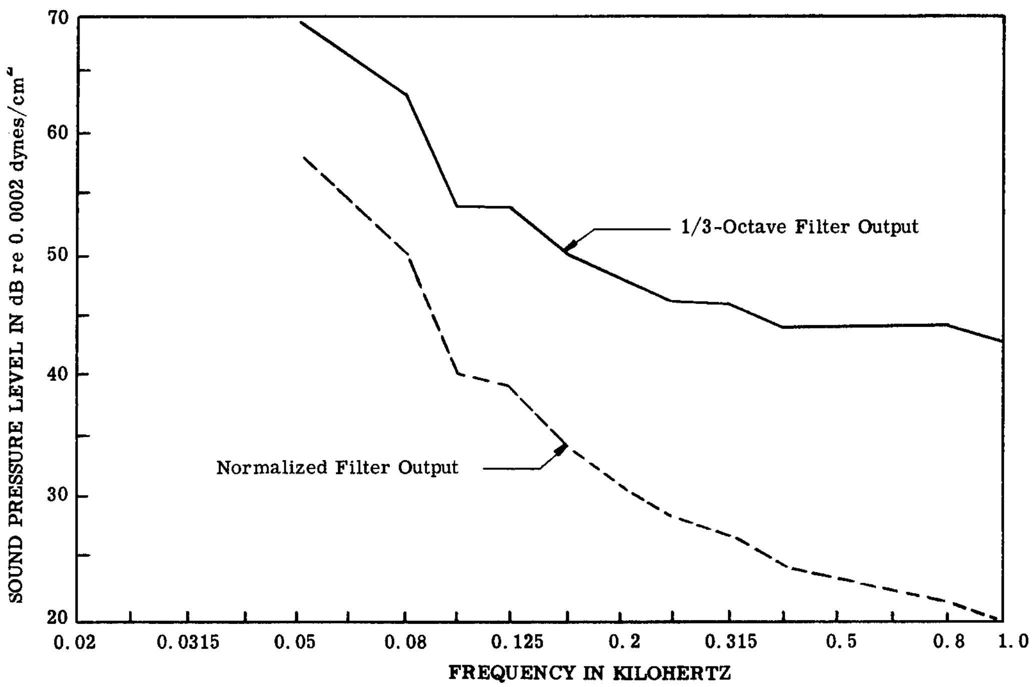


Figure 2-10. One-Third-Octave Analysis of a Continuous Noise Source Showing the Effects of Bandwidth Normalization

signal level can be calculated from the following expression (for whose derivation see, for example L. L. Beranek, Acoustic Measurements, John Wiley and Sons, New York (1949), pp. 563-564.

$$C = -10 \log (f_a - f_b), \text{ dB} \tag{37}$$

where

f_a = upper cutoff frequency

f_b = lower cutoff frequency

The bandwidth correction method described above assumes that the signal spectrum being measured is roughly constant over the bandwidth of the filter element and, if a voltage detector rather than a power detector is used, that the phase information across the band is random. This assumption is generally of sufficient accuracy to cause no concern in the analysis of signals whose spectra are continuous. The signal

must vary rather abruptly with frequency to change the conclusions which would result from this simple procedure. The problem associated with this procedure and its application to signals having periodic components lies in the fact that the spectrum of an individual line may be narrower than the filter applied to the data. In this instance, the amplitude of the line will not be a function of filter bandwidth until the filter is made narrower than the equivalent bandwidth of the line. For example, a narrow-line component whose power is distributed over a spectral band equal to 1% of its center frequency will have the same amplitude when viewed through a 1/3-octave filter or a 1/10-octave filter.

c. Phase and Amplitude Coherence. In addition to the abovementioned considerations of using a 10 log bandwidth correction for normalization of spectral levels, a complexity occurs with signals which tend to have phase and amplitude coherency across the analysis bandwidth. One example of such a signal would be that of an explosion which results in a frequency amplitude and phase behavior fairly consistent over a given analysis bandwidth. For this case the behavior of the peak signal voltage level will decrease by 20 log bandwidth versus 10 log bandwidth as the analysis resolution is diminished. Generally signals which are transient in nature will behave this way, and require care in determining the meaningfulness of the normalized spectral output. In real life, such transient signals will behave with the peak measured amplitude level decreasing between 10 and 20 log bandwidth depending on the degree of the amplitude and phase consistency and the type of detector employed. A true power detector will measure the power level regardless of phase coherency and so a 10 log bandwidth correction is applicable (subject to the "narrow line" problem discussed below). When voltage detectors are used, however, the proper correction will be a function of phase coherency in the band. Peak voltage detectors are frequently employed in the analysis of transients because of the difficulty of properly averaging a short duration signal.

In a practical sense, there are two methods of dealing with this situation which have been applied in analysis. The least troublesome technique is to apply the simple filter bandwidth correction to all data and to explain the amount of correction and its meaning in the accompanying text. This procedure allows the signature user to consider the possible effects such a change would have on this system and he may remove any corrections which may confuse his data interpretation.

A somewhat more accurate picture of the complex spectrum can be obtained if a closer look is taken at this process of bandwidth normalization. In particular, it should be remembered that the normalization is a means of accommodating the scale of amplitude (voltage,

pressure, etc.) versus frequency graphs to a common 1 Hz bandwidth basis for comparative purposes. Thus normalization may be carried out in terms of a filter correction factor only in those cases where the signal spectrum is reasonably constant across the filter. Narrowline components cannot be corrected in this manner. It is possible, however, to correct narrowline components if an estimate can be made concerning their width.

Such an estimate can sometimes be made by reanalyzing the data with a narrow-band analyzer. Frequently, it can be established that the spectral widths of line components are not infinitesimal but may in fact be, say 1% of their center frequency. If such an estimate can be made, the filter output (regardless of its width) can be divided by the equivalent bandwidth of the line component to achieve the correct level. Obviously, however, the location of the line is in doubt within the filter width. Should the analysis filter have a negative S/N for the line component as its output, a narrower analysis bandwidth must be used for the initial analysis. This correction procedure produces a more accurate estimate of the spectrum of a complex signal on a per cycle bandwidth basis, which is compatible to the "spectrum levels" normally used for noise surveys. Ideally, one would prefer to analyze the data with a 1-Hz resolution to get "spectrum levels" directly. The primary problem associated with the above procedure is the documentation necessary to eliminate any confusion on the part of the user as to what exactly has been done to produce the data. Often, the user would prefer to have the original analyzer data which he can interpret in the light of his experience with the particular analyzer.

The effect of these corrections can be seen in Figures 2-11 and 2-12. Figure 2-11 is a spectrum of a complex source, as measured with a constant bandwidth (approximately 20 Hz) analyzer, with correction for the filter bandwidth and for line components whose width is assumed to be 1% of their center frequency. It is further assumed that the energy in this line is evenly distributed within the 1% bandwidth and the amplitude of the line component is at least equal to the noise in the filter band being corrected. Under these assumptions the correction for line frequencies increases with frequency, while the correction for the random noise in between the lines is a constant 10 log 20=13 dB. The correction for the 50 Hz line, which is assumed to have a 0.5 Hz bandwidth is zero, since it is contained within the 1 Hz bandwidth.

Figure 2-12 is a spectrum of the same source as measured with a 1/3-octave analyzer. In this case, the uncorrected simple filter bandwidth and individual line component corrections are all shown. The effects of the corrections are reasonably obvious and should be considered in light of the expected noise environment. If the noise sources are similar to the signal (i.e., both continuous or both

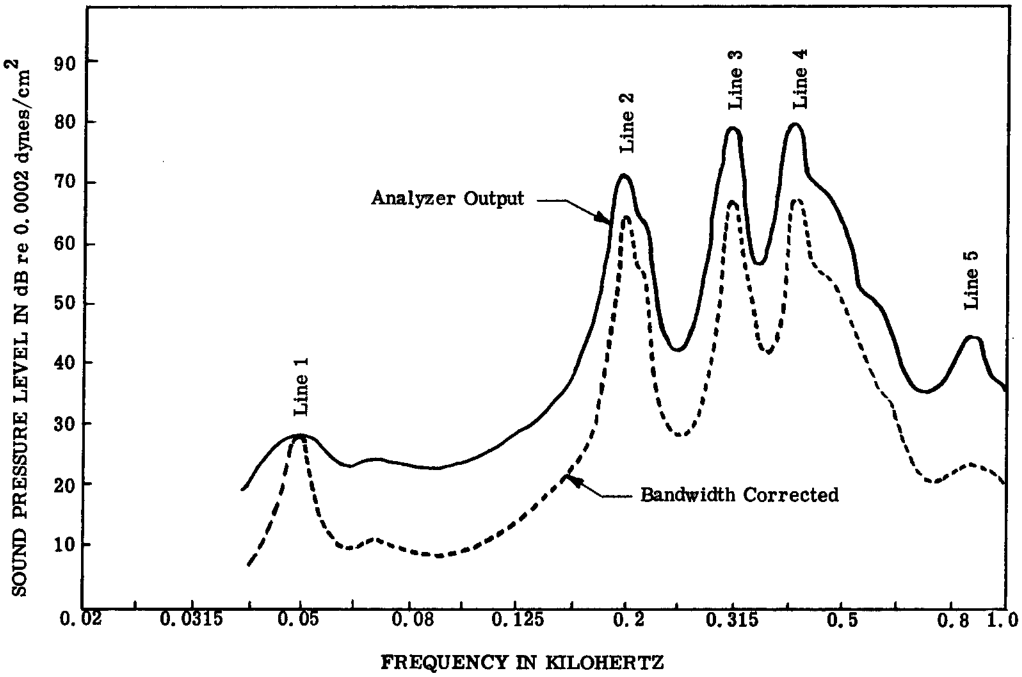


Figure 2-11. Spectrum Analysis of a Complex Signal Using a Constant Bandwidth (20 Hz) Analyzer. Broken-Line Curve Shows Effect of Bandwidth Correction Assuming a Uniform Line Width of 1% of Its Center Frequency

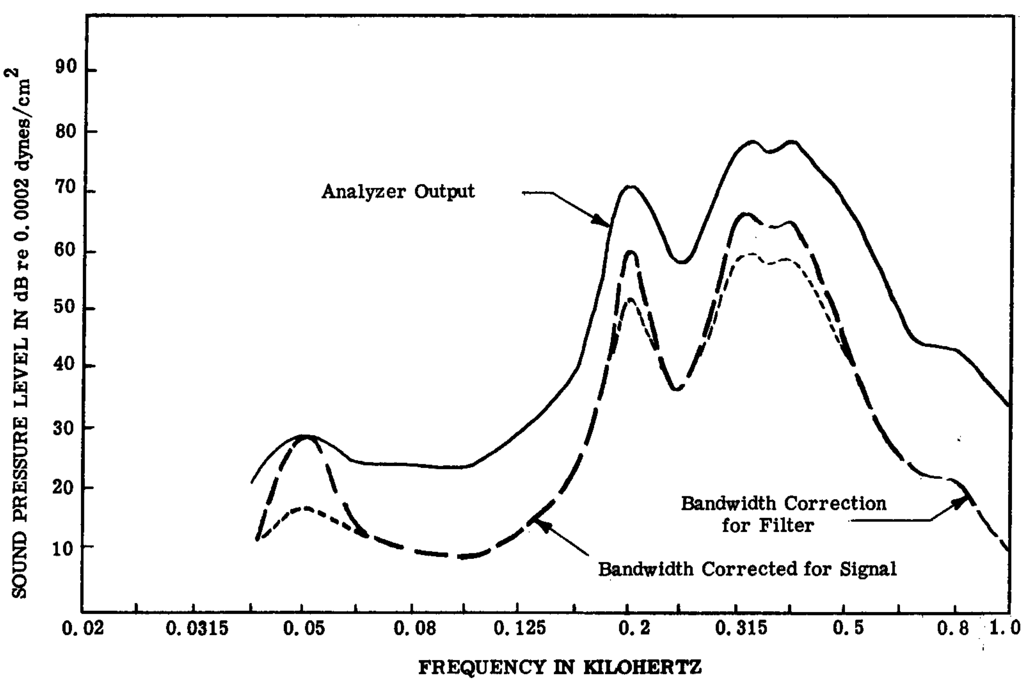


Figure 2-12. Spectrum Analysis of a Complex Signal Using a 1/3-Octave Analyzer. Large Broken-Line Curve Shows Effect of Bandwidth Correction Assuming a Uniform Line Width of 1% of Its Center Frequency. Dotted Portion Shows What the Corrected Graph Would Have Been if Simple Bandwidth Correction Had Been Used.

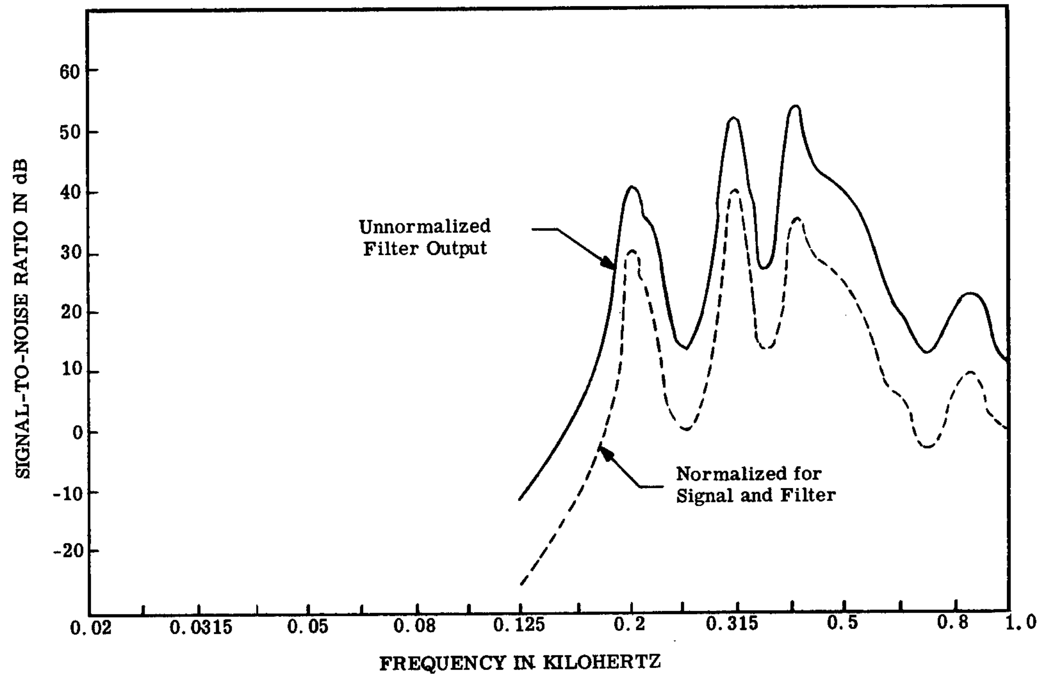


Figure 2-13. Signal-to-Noise Ratio Estimated for the Combination of Noise Shown in Figure 2-8 and Complex Signal Shown in Figure 2-11

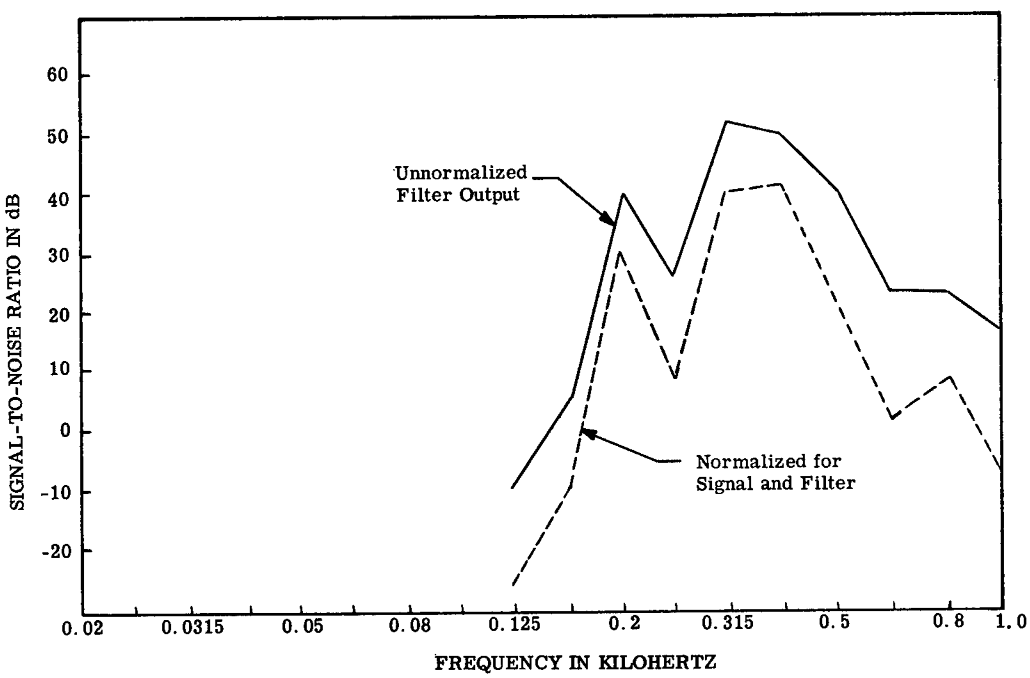


Figure 2-14. Signal-to-Noise Ratio Estimated for the Combination of Noise Shown in Figure 2-8 and Complex Signal Shown in Figure 2-12.

2.3.A

complex with line structure), the superposition of their signatures is a useful measure of the expected S/N regardless of the method used, as long as it is the same for both.

Figure 2-13 shows a typical signal-to-noise plot for the continuous noise source of Figure 2-11 and the complex signature presented in Figure 2-10. We see the expected S/N with and without corrections. Likewise, the same noise source is compared to the spectrum derived from the 1/3-octave analysis (Figure 2-12) and the resulting corrected and uncorrected S/N ratios are shown in Figure 2-14.

A word should be said about the units used in acoustical data analysis. In general, the levels may be reported as relative levels or absolute levels. Usually, the acoustic reference level is 0.0002 microbar (dyne/cm^2) in air, and 1 microbar in water.

The level of sound pressure measured with a given filter bandwidth is referred to as Sound Pressure Level (SPL) in (1/3-octave, etc.) bandwidth, or Band Pressure Level (BPL). When the analysis bandwidth is 1 Hz, or when normalized to 1 Hz bandwidth as described above, it is referred to as Pressure Spectrum Density Level, or simply Pressure Spectrum Level. A commonly used instrument, the General Radio Type 1564-A Analyzer is continuously tunable from 2.5 Hz to 25 kHz in four ranges. A 1/3-octave band and a 1/10-octave band are provided. Charts for converting from the measured band levels to the spectrum level are given in the Handbook of Noise Measurement, General Radio Company, by A. P. G. Peterson and E. E. Gross, Jr.

4. Calibration of Spectral Analysis Measuring Equipment

The most widely used method of calibrating frequency analysis equipment for effective bandwidth Δf is to process a band-limited white noise of known total bandwidth F and a sine wave of adjustable frequency f. The analyzer is tuned to maximum reading with a given sine wave input of given frequency and voltage E applied to the input terminals. The power input then is

$$P = E^2/R \qquad (2\text{-}38)$$

where R is the analyzer input resistance. The analyzer rms output voltage indicator is adjusted to a reference value. Next, white noise of rms voltage E is applied to the input terminals producing the same input power P, but resulting in an indication which is a G-fraction of the reference value. Then, the effective noise bandwidth of the analyzer at the frequency f is

$$\Delta f_f = G^2F \qquad (2\text{-}39)$$

The "whiteness" and the bandwidth of the noise source can be verified with a constant bandwidth analyzer.

Usually, the spectrum is displayed on a logarithmic scale, in dB, in order to show more detail for the low power components The accuracy of the logarithmic amplifier can be measured by varying the amplitude of the calibration signal, usually in discrete steps corresponding to a fixed number of dB.

B. Constant Bandwidth (Δf) Analysis

1. Δf Instrumentation

The constant bandwidth systems are so named because their analysis bandwidth Δf is invariant with frequency. However, most of these analyzers have several bandwidths available which can be selected by the operator and which remain constant at all frequencies.

The constant bandwidth analyzers commonly use a heterodying approach to synthesize a narrow constant-bandwidth filter which is swept through the passband. In these systems (Figure 2-15) the original signal is caused to modulate a sine wave source (carrier) in a balanced modulator. The carrier is suppressed by the balanced modulator so that the output consists only of two sidebands symmetrically placed about the carrier frequency at frequencies $\omega_c \pm \omega_o$ where ω_c is the carrier angular frequency and ω_o is the intelligence angular frequency (Figure 2-16). The amplitude of the sidebands is proportional to the amplitude of the input signal and to the amplitude of the carrier. The latter is held constant so that the amplitude of the sidebands is proportional only to the amplitude of the input signal. The effect is to translate the original output spectrum to a higher frequency region. A narrow-band fixed bandpass filter at ω_f is used to select the signal elements within $\pm\ \omega_d$ of ω_f where ω_d is the half-angular bandwidth of the filter. Usually, quartz crystal filters are used for stability and high-Q and bandwidths of 1, 10 and 100 Hz are common. The output of the filter is detected, filtered, and displayed. The effect is to synthesize a filter of half-angular bandwidth ω_d at ω_o where

$$\omega_o = \omega_c - \omega_f \tag{2-40}$$

where ω_c is the carrier frequency and ω_f is the fixed filter frequency.

By varying ω_c the effective filter frequency ω_o is correspondingly varied. Eq. (2-40) applies to the case here the lower sideband is selected by the filter. This is the typi-

2.3.B

cal approach since it maximizes the bandwidth which can be analyzed at a given filter frequency without aliasing.

An input low pass filter is customarily used to prevent aliasing which occurs if the input frequency region extends into the lower sideband region. This would happen if the input spectrum contained components above one-half the lowest carrier frequency. In the case where the lower sideband is selected for analysis, the lowest carrier frequency equals the filter frequency, i.e., $\omega_c = \omega_f$ for a synthesized filter at $\omega_o = 0$.

Actually, a filter is seldom synthesized at $\omega_o = 0$ since imperfect cancellation of carrier (always a practical problem) would pass through the filter and contaminate the reading. There is a lower limit to the frequency at which one can synthesize the filter since, for low frequencies, the upper and lower sidebands (as well as carrier leakage) are close together and the finite slope of the bandpass filter cannot reject the unwanted components. It is possible to use the phase-shift method of single sideband generation to suppress the unwanted sideband by say, 40 dB and, with extra care in carrier suppression, the lower limit of the device can be extended.

Typical specifications for a constant bandwidth analyzer of this type are:

Input Range	--	20 Hz to 54 kHz
Bandwidth	--	3, 10, and 50 Hz
Filter Frequency	--	100 kHz
Carrier Frequency	--	100.020 kHz to 154.00 kHz

The advantage of these schemes is the ability to have a single filter which can be made very near ideal. The bandwidth of such a tuned network can be made very narrow since a constant-frequency tuned circuit is used which can be designed to have good selectivity. This gives the necessary resolution for the measurement of very narrow line components in the data. The SSB technique described is useful to relatively low frequencies (as low as 2 Hz) when the degree of carrier suppression can be made quite good.

An example of Δf analysis is shown in Figure 2-17.

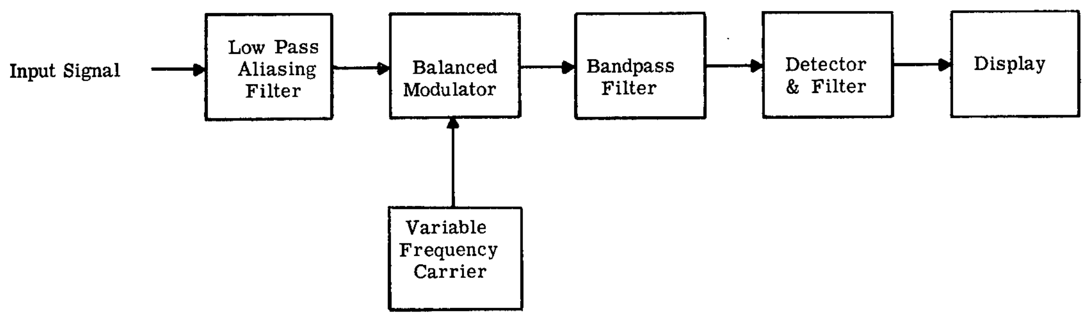


Figure 2-15. Block Diagram of Typical Constant Bandwidth Analyzer

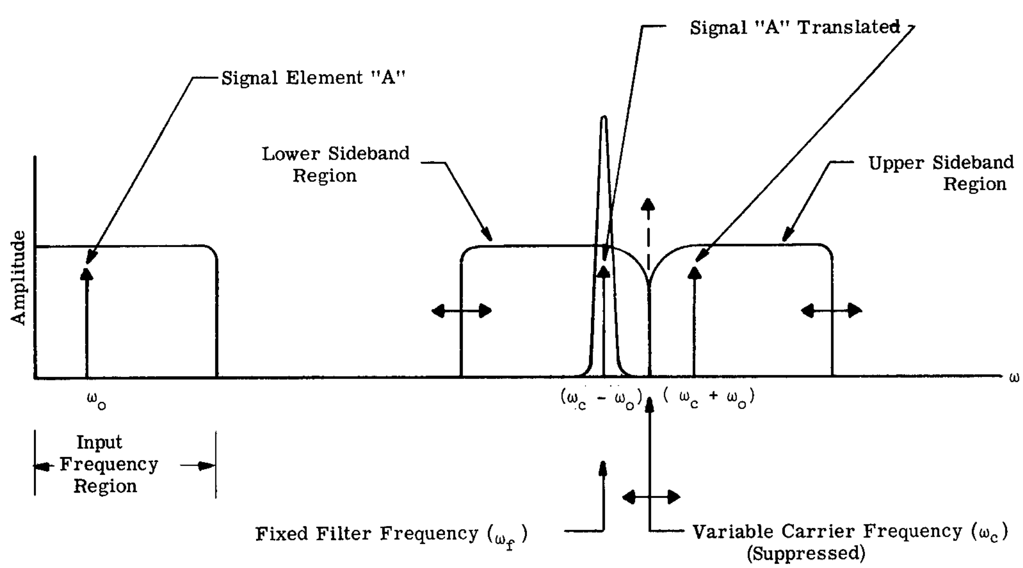


Figure 2-16. Spectrum Translation

2. Interpretation and Application

Constant bandwidth analyzers are most useful in identifying line components in a signal especially when the lines are closely spaced. Because of the high resolution of the analyzer, multiple line structures are resolved. By using a 1 Hz resolution bandwidth, pressure spectrum levels (PSL) are automatically generated.

The key advantage of the swept-constant-bandwidth filter is its ability to maintain narrow and constant bandwidth over a wide frequency range--for example, a 3 Hz resolution from 20 Hz to 54 kHz. No other practical analyzer has this feature. The main disadvantage of the swept-constant-bandwidth analyzer is that it generates only one filter function at a time, and the filter must be swept or tuned to different parts of the band. To perform a complete analysis, the same sample of data must be repeated continuously by recording it on a magnetic tape loop. The filter is incremented one resolution bandwidth for each pass of the tape. Thus, it requires many passes and, consequently, a long time to analyze one segment of data.

For example, assume that a resolution bandwidth of 1 Hz is desired and the frequency band of interest is from 20 Hz to 5000 Hz. In order for the 1 Hz-filter to respond to the data sample, it must be connected for approximately 1 second (the reciprocal of the bandwidth). Thus, the minimum sample repetition rate is once per second. If the filter is incremented 1 Hz/sec, it will require 4980 seconds or 83 minutes to analyze one second of data! This is hardly an economical approach and so the swept-constant-bandwidth narrow-band analyzer is generally used only to find and measure specific line components. It is not a practical tool for complete wideband analysis.

C. Proportionate Bandwidth ($\Delta f/f$) Analysis

1. $\Delta f/f$ Instrumentation

Proportionate bandwidth analyzers are so named because their analysis bandwidth varies directly with frequency, i.e., as the center frequency of the analyzer increases, the bandwidth increases proportionately, so that

$$(\Delta f/f) = \text{constant} \tag{2-41}$$

where Δf is the filter bandwith

f is the filter center frequency

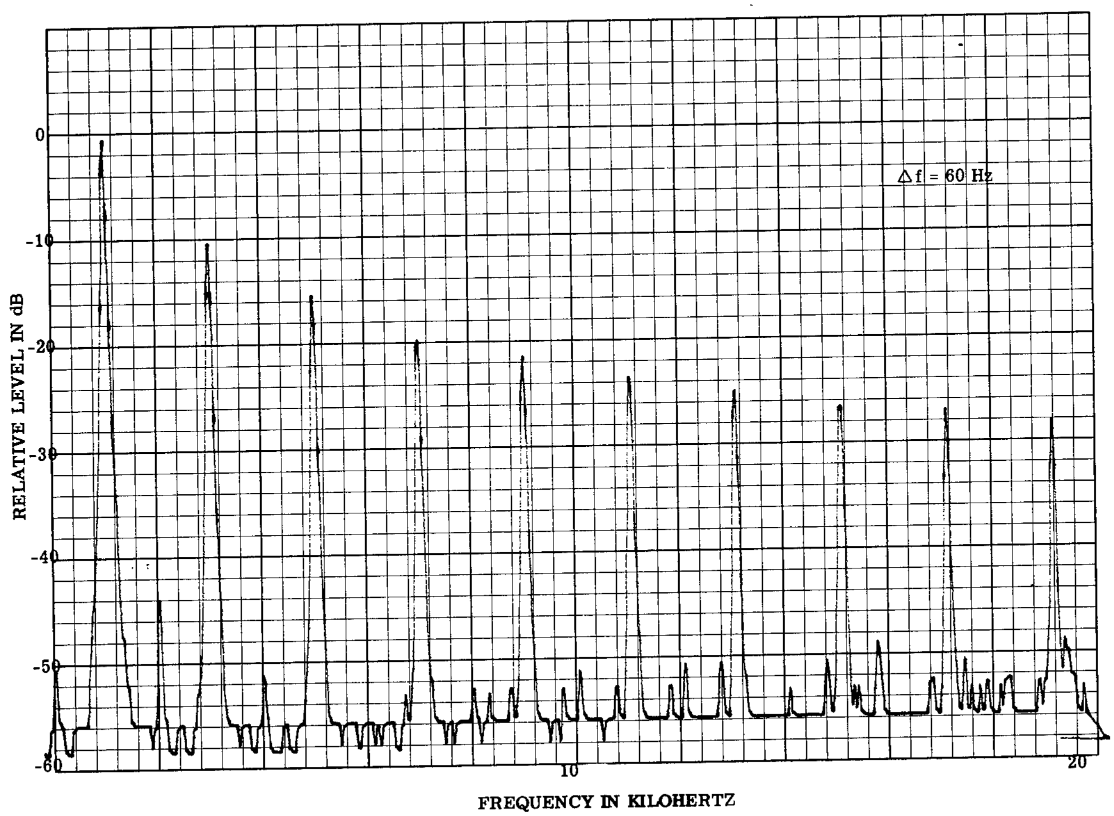


Figure 2-17. Δf Analysis of 1 kHz Square Wave

2.3.C

The bandwidth is thus a constant fraction of an octave. Typical proportionate bandwidth analyzers have filter bands of 1/10-octave, 1/3-octave and one octave, although 1% and 10% wide filters are also in common use.

a. Contiguous Band Analyzers

Proportionate bandwidth analyzers can be divided into two categories, parallel or contiguous band analyzers and swept analyzers. A contiguous band analyzer consists of a bank of filters whose inputs are driven in parallel (Figure 2-18). In general, these filters intersect at their half-power points and have bandwidths which are a constant percentage of their center frequency (Figure 2-19).

This type of analyzer can continously display all of the spectral components on a multichannel oscillograph, thus assisting rapid data reduction. Such a multichannel data display is especially useful for examination of the time characteristics of the many portions of the spectrum. This aspect is especially important if the signal is non-stationary. Some of the available analyzers apply commutation to the output of the filter bank so that a single detector can be used (Figure 2-20). This destroys some of the above advantages.

The primary disadvantage of the contiguous band analyzer is the cost of a large number of filters with sufficient skirt selectivity to enable wide dynamic-range signals to be resolved.

b. Swept Band Analyzer

The second category of proportionate $\Delta f/f$ bandwidth analyzers is the so-called "swept" analyzer (Figure 2-21). As in the swept-constant-bandwidth analyzer, a single filter is swept through the band of interest. Here, however, the filter bandwidth is a constant fraction of the center frequency. This type of performance is obtainable with variably tuned L-C filters. The swept proportionate bandwidth analyzer maintains one of the main advantages of contiguous proportionate bandwidth filters in that the system resolution is independent of frequency. This type of analyzer offers several realistic advantages, such as low cost, small size, low power consumption, and simplicity of operation. These improvements are gained at the expense of the continuous readout of the entire spectrum which the contiguous band analyzer is able to provide, resulting in diminished ability to analyze non-stationary signals.

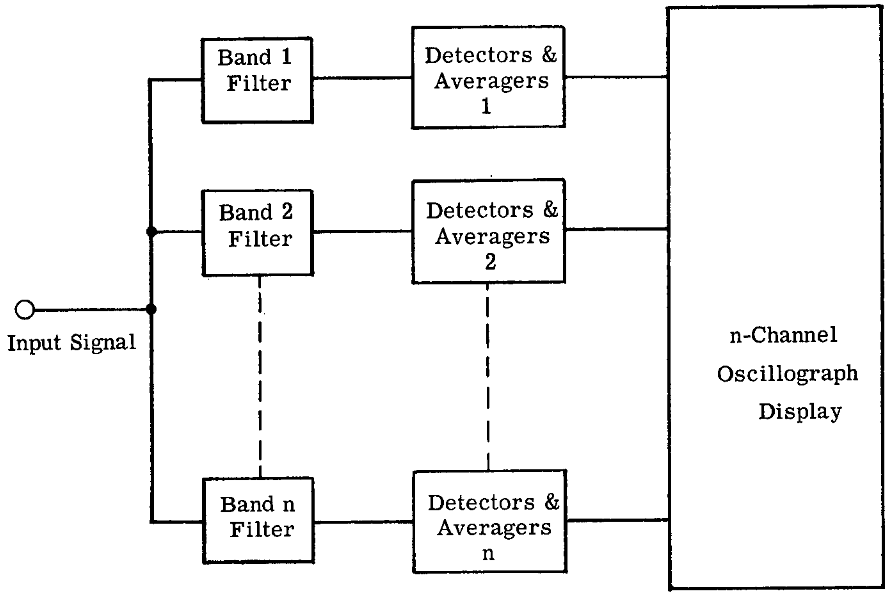


Figure 2-18. Parallel or Contiguous Band Analyzer

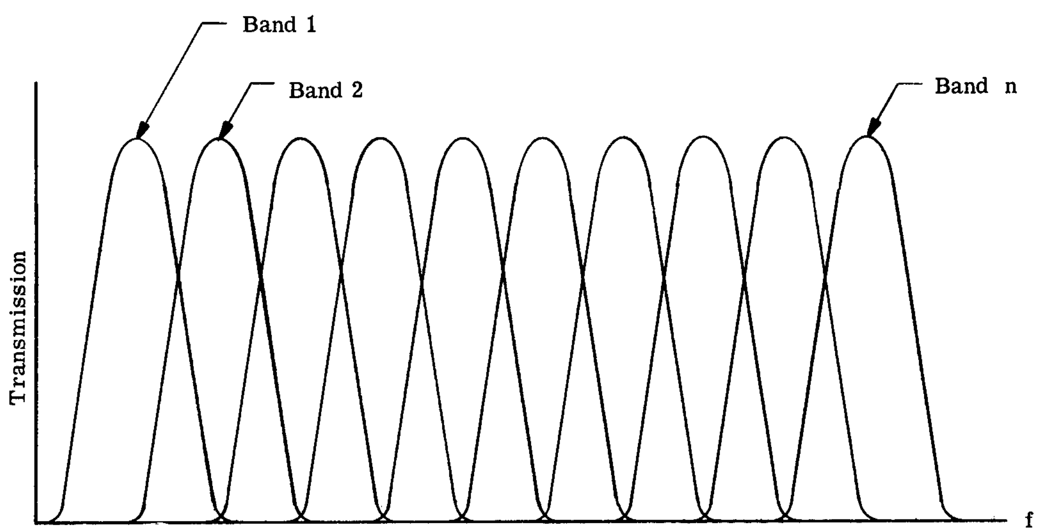


Figure 2-19. Contiguous Band Analyzer Filter Response

2.3.C

Since only a single filter and detector are used, a sample of data is normally recorded on a magnetic tape loop and repeated as the filter center frequency is moved. Since the bandwidth of the filter varies directly with frequency, the filter can be swept at a constant number of octaves per unit time. This greatly speeds up the analysis. For example, a 1/3-octave analyzer can be swept at a rate of roughly one octave every 3 seconds. In the band from 20 Hz to 5000 Hz, there are approximately 8 octaves. The analysis of such a band can be performed in about 24 seconds using a 1/3-octave analyzer or 80 seconds using a 1/10-octave analyzer. Nevertheless, since the analyzer does not operate in real time, one cannot get a continuous readout on non-stationary signals. Further, since the analysis time is constant, the time-bandwidth (BT) product varies with frequency. Thus, the statistical accuracy of the analysis varies, being better at higher than at lower frequencies. Of course, since the bandwidth is greater at higher frequencies, one knows less about the exact frequency of each component, even though one knows the band energy with greater accuracy. An example of $\Delta f/f$ bandwidth analysis is given in Figure 2-23.

c. <u>Swept Band Analyzer with Integration</u>

An interesting adaptation of 1/10-octave swept filter analyzer, which includes a 1 second integration, has been used for some AWG studies. Figure 2-22 is a block diagram of the analyzer and integrator. The signal to be analyzed is recorded on a loop tape recorder. The loop is marked (for example, with a piece of reflecting foil) to indicate the "start" of the loop. This start mark is used to synchronize the subsequent integrator. The signal is filtered by a 1/10-octave sweeping analyzer which slowly scans through the desired range. The scan rate is kept well below 1/10-octave per second. The output of the analyzer is rectified and integrated in a "true" manner by charging a capacitor without leakage resistor. The time for integration is 1 second, as determined by the length of the tape loop. When the "start" mark is reached, any capacitor charge is transferred to a holding circuit and the integrator is returned to 0 in preparation for the next pass. Meanwhile, the transferred charge is displayed on a graphic-level recorder chart. A balanced modulator is used to convert the average power level to a form useable by the graphic-level recorder. At the end of the pass, the charge on the capacitor again is transferred and displayed. Thus, the chart record indicates the energy level in the 1/10-octave band, i.e., the power level, integrated over a 1-second period.

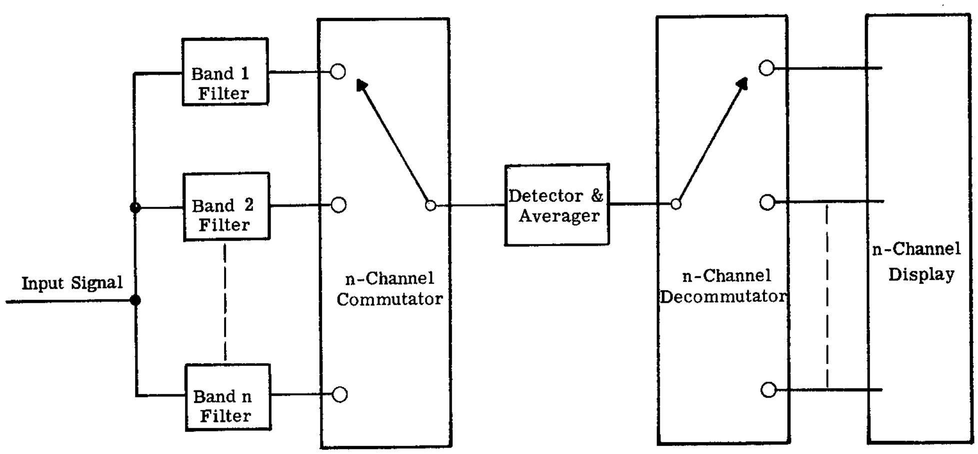


Figure 2-20. Contiguous Band Analyzer with Commutated Detector

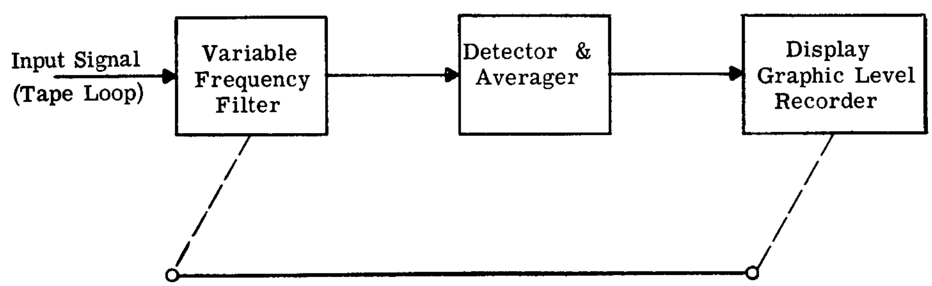


Figure 2-21. Swept Proportionate Bandwidth Analyzer

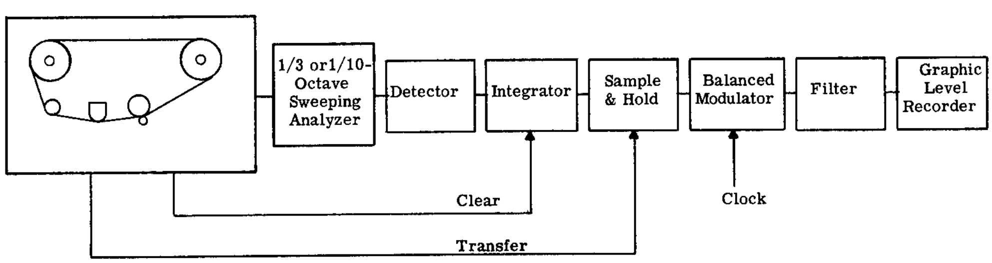


Figure 2-22. Block Diagram of Analyzer and Integrator

2.3.C

By repeating a signal every T seconds, a line series is generated based on a fundamental frequency, 1/T, and its harmonic components. By choosing a repetition rate of 1 second, the line spectra is based on a 1 Hz fundamental. Analysis of random phenomena with a 1/10-octave filter produces a 1/10-octave band sound pressure level plot. When analyzing transient signals, which take place during a fraction of the loop length, an approximation to true pressure spectrum level is obtained. Analysis of such transient signals is helped by using a "window" to isolate the transient phenomena from background noise. This "window" is synchronized in time from the "start" mark of the tape. It is used to gate only the desired portion of the tape loop into the analyzer. Surrounding background noise is thus minimized while maintaining the 1-second repetition rate. Strong single-line signals result in readings directly indicative of their SPL.

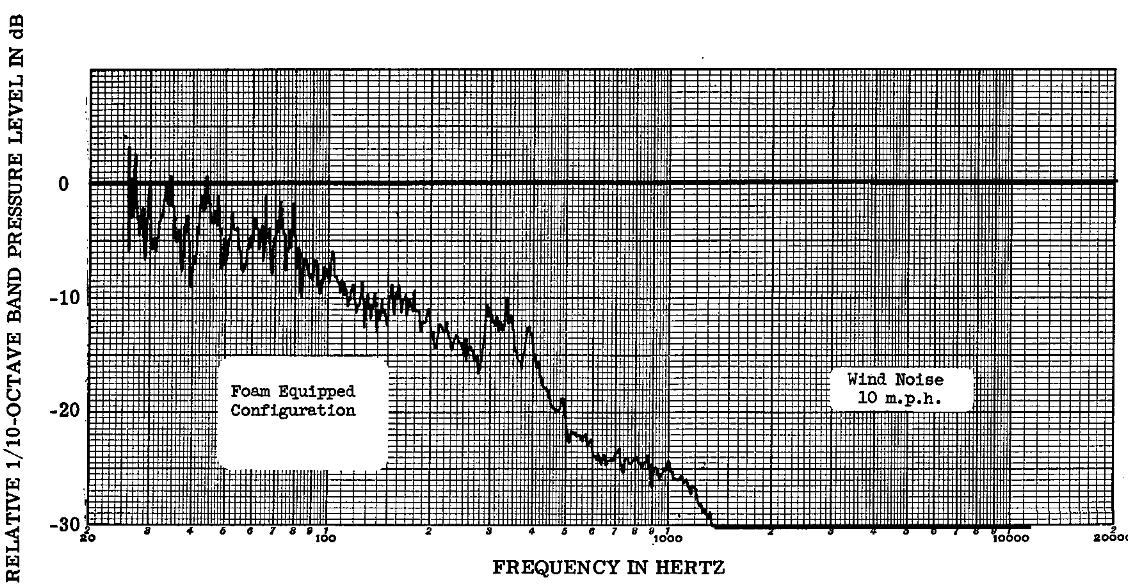


Figure 2-23. Example of Δf/f Analysis of the Noise Generated by a 10 mph Wind Speed on a Microphone

2. Interpretation and Applications

Care must be taken in interpreting the data from proportionate bandwidth analyzers. Since the filter bandwidth is not constant, but increases with increasing center frequency, the statistical accuracy for a given analysis time is greater at high frequencies than at low when examining broadband signals. Further, when such an analyzer is used to examine "white" noise, i.e. noise with an equal energy density per cycle, the indicated output or band pressure level increases at a 3 dB/octave rate corresponding to the doubling of the filter bandwidth every octave. Based on this, some analysts impose a 3 dB/octave negative slope on either the input or output data to compensate for the variable bandwidth. It should be remembered that such a correction is valid only for white noise which is only approximated in practice by random phenomena. Furthermore, any frequency analyzer will yield the correct value for an isolated line component directly since, if there is no other component within the filter bandwidth, the output is independent of the filter bandwidth. In this case, no correction for filter bandwidth should be applied. The corrections, if any, to be applied in particular instances are discussed more fully in subsection 2.3.A.3.

The relatively poor resolution of the proportionate bandwidth analyzer can either be an advantage or disadvantage depending upon the data sought. For example, if the signal to be analyzed consists of relatively widely spaced line components with frequency instability, the wider bandwidth of the proportionate bandwidth analyzer can encompass the line even as it dithers. Thus, the output of the analyzer is proportionate to the total energy of the line. Of course, no information is gleaned about the dither rate or even if dither exists. On the other hand, if two lines fall into the same filter bandwidth, they cannot be separated and their relative strengths remain unknown. Also, if a line component is buried in noise, the relatively wide bandwidth does not allow one to ascertain the existence of the line.

From the above, it can be seen that no single method of analysis is able to provide all aspects of data that may be of interest. Some *a priori* knowledge of the signal composition is helpful in deciding on the best approach to analysis and in subsequent interpretation of results. A trained human ear is often the best preliminary guide as to the method of analysis which is apt to be the most appropriate in any particular case.

D. Real-Time Analysis

1. Real-Time Instrumentation

In general, the previously described frequency analyzers do not have the capability of performing the analysis in "real time," that is, keeping up with a continuously changing non-stationary series. The real-time analyzer is similar to the constant-bandwidth-swept-filter analyzer described in subsection 2.3.B.1; however, the data sample is stored in a digital delay line and is rapidly recirculated. This translates the data up in frequency so that a rapid analysis can be performed. The stored data is continuously updated by dropping one element in time and adding another as the circulation proceeds, so that, for all intents and purposes, the analysis is carried out piecewise on overlapping frames of a series which differ from each other by a predetermined small-time interval. The resulting spectrum is customarily displayed on an oscilloscope screen with frequency and spectral intensity information along the horizontal and vertical deflection co-ordinates, respectively. Also, any desired number of displays can be summed for the purpose of obtaining an average over a number of frames, stored in an accumulator, and either displayed on the oscilloscope, or graphed with an X-Y plotter.

The detailed operation of a real-time analyzer is shown in Figure 2-24. The input signal is low-pass filtered to prevent aliasing in the subsequent sampler. The data is then sampled at a rate at least twice the highest frequency transmitted by the filter. Usually, a sampling rate of three times the upper frequency is chosen. The samples are then converted into digital words whose length is commensurate with the accuracy required. Typically, a nine bit word is used. This provides quantization to 512 levels for a dynamic range of 54 dB. The digital words are stored in a delay line memory and recirculated at a rate such that the entire data stream in the line is passed once between each sample period of the input signal. As each new sample of the signal enters the delay line, it replaces the oldest data sample in storage.

The output from the recirculating delay line is converted to an analog signal by a high-speed D-A converter. This high-frequency analog signal is analyzed using a heterodyning method similar to that described in subsection 2.3.B.1. That is, the high-speed signal is multiplied by a local oscillator frequency. The sidebands are then passed through a fixed narrow bandwidth crystal filter which selects the lower sideband for detection. The local oscillator signal is stepped through the range of interest by the master clock of the analyzer.

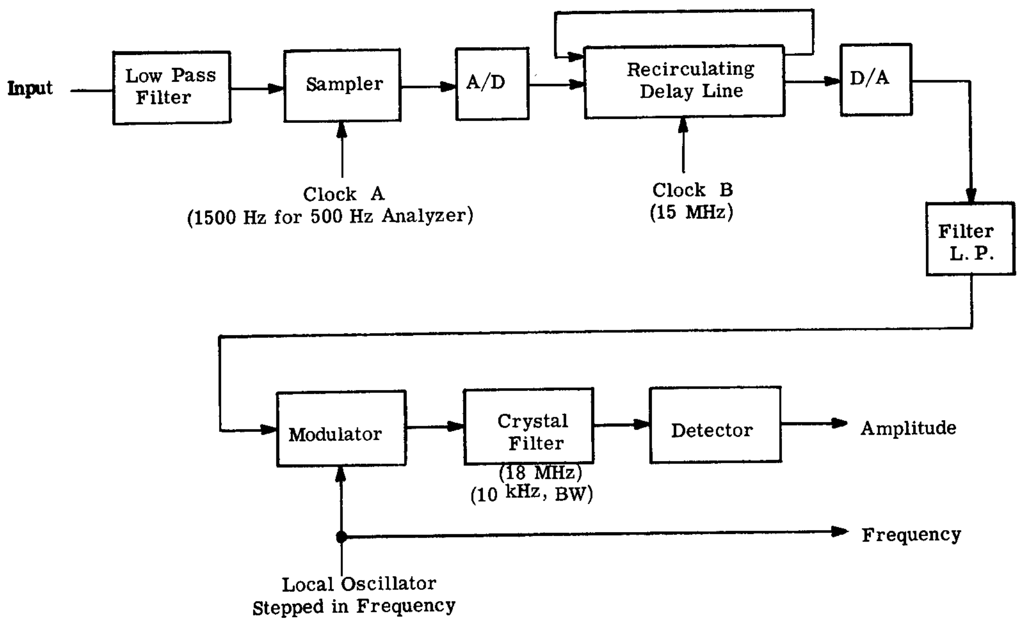


Figure 2-24. Simplified Block Diagram of Real-Time Analyzer

2.3.D

Real-time analyzers are categorized by the "number of line analysis" they are capable of performing. This quality factor is a function of the amount of storage in the delay line. For example, a 500 point analyzer is capable of synthesizing, in effect, 500 parallel filters throughout the band of interest. Normally, the bandwidth of the filters and the corresponding analysis bandwidth are selectable by switch; however, the bandwidth which can be handled will always be 500 times the resolution bandwidth. Such a device may provide for a 1-Hz analysis bandwidth up to a highest frequency of 500 Hz. The 500-Hz signal will be sampled at a rate of 1500 samples per second. In order to provide the 1-Hz resolution bandwidth, the sample duration in the recirculating delay line must correspond to 1 second of real time. Thus, 1500 samples are stored in the delay line at any given time. Alternately, the instrument could have been set to handle a data bandwidth of 5 kHz with a 10-Hz resolution. In this case, the sample duration which would have filled the delay line would have been 100 milliseconds and the total number of points would again be 1500.

In order to minimize the errors due to the non-infinite data sample, the real-time analyzer generally applies some sort of window weighting function to the data. This concept was covered in subsection 2.2.E.4. Typically, cosine-squared weighting is employed in a real-time analyzer. An unfortunate result of the weighting function is to increase the effective bandwidth of the synthesized filters. For cosine-squared weighting, this amounts to about 50% That is, a 500-line analyzer set to a 500-Hz bandwidth synthesizes effective filter bandwidths of 1.5 Hz rather than 1 Hz.

2. <u>Accumulator and Averager</u>

A useful adjunct to the real-time analyzer is the companion integrator or averager. This device determines the average spectral level over a given number of ensembles. In a typical instrument, it can be set to average the results of from 2 to say 1024 spectral analyses. This is a very powerful tool for detecting line structure in the presence of noise. Averaging the data over a period of time improves the time bandwidth product and, thus, the statistical accuracy of the results. In general, one should average over the largest number of ensembles possible consistent with the stability of the data.

The typical averager used with a real-time analyzer accumulates the data in a second recirculating digital delay line. As the data from each of the effective filters enters the averager, it is digitized and placed in the delay line. The recirculation rate of the delay line is synchronized with the data flow rate from the analyzer, so that each filter has its own section or

2.3.D

"bin" of delay line. With every recirculation of the data, the new output is added to the previous value in each bin, the sum being representative of the average. The data can then be read out on any suitable display. Examples of real-time analysis of truck data are shown below in Figures 2-25 and 2-26.

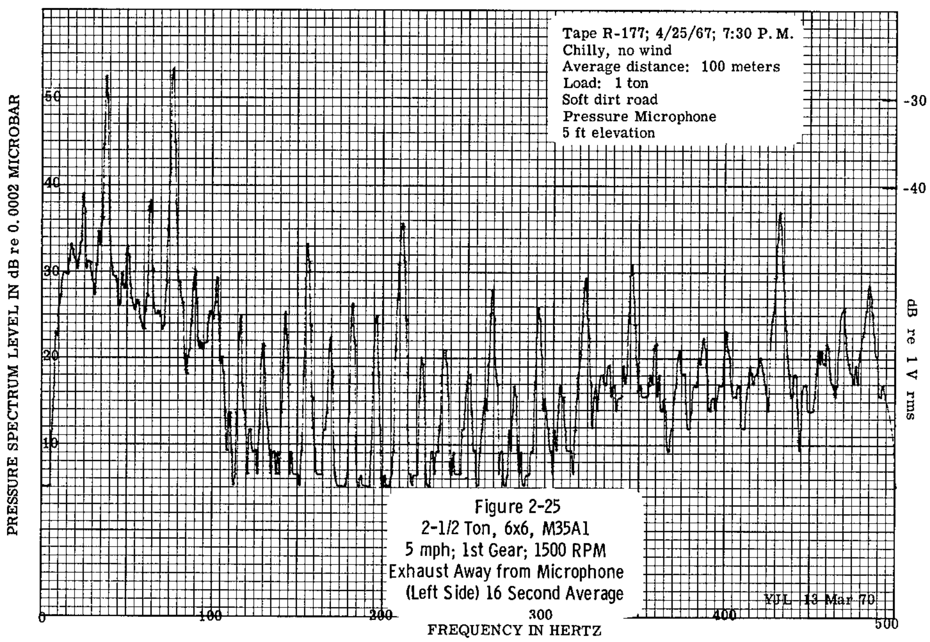


Figure 2-25
2-1/2 Ton, 6x6, M35A1
5 mph; 1st Gear; 1500 RPM
Exhaust Away from Microphone
(Left Side) 16 Second Average

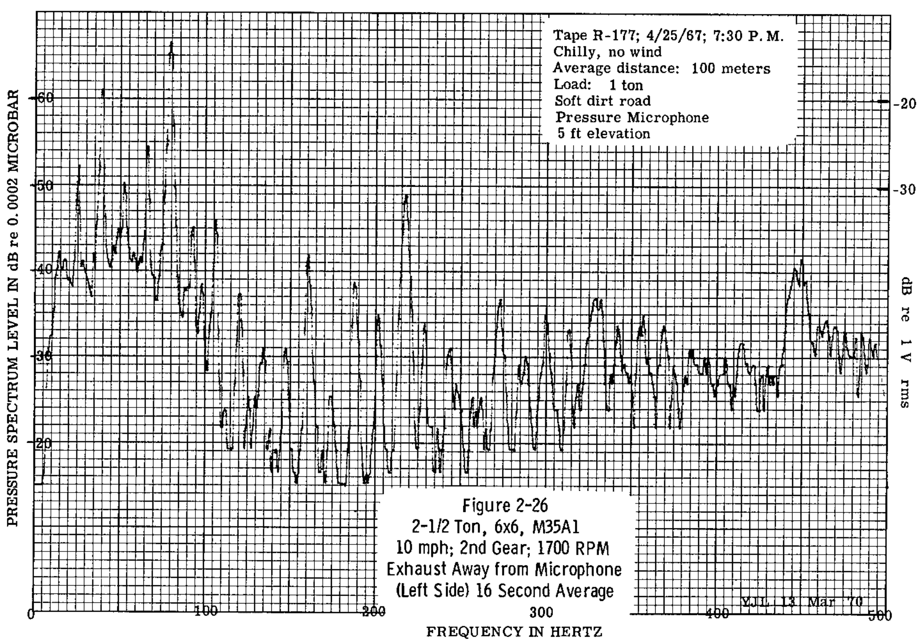


Figure 2-26
2-1/2 Ton, 6x6, M35A1
10 mph; 2nd Gear; 1700 RPM
Exhaust Away from Microphone
(Left Side) 16 Second Average

3. Interpretation and Applications

The spectra produced by a real-time analyzer are very similar to those of the constant bandwidth analyzer of subsection 2.3.B, since the real-time analyzer is a constant bandwidth device. The key difference between them is that the constant bandwidth analyzer must be swept through the band of interest at such a slow rate as to make it impractical for complete broadband analysis. The real-time analyzer, on the other hand, synthesizes the effect of a parallel filter bank of constant bandwidth filters. Thus, it processes the data in real time and the analyst can observe the stationarity of the analysis. Alternately, he can accumulate and average the spectrum produced over a period of time.

The drawback of the real-time analyzer is the number of parallel filters it can synthesize. This is directly related to the cost of the device. Typically 200- and 500-point analyzers are available. One thousand-point analyzers also have been made for special purposes.

Because the total number of filters is fixed, the normal way to increase the input bandwidth which the analyzer will accommodate is to widen the analysis bandwidth. The analysis bandwidth is given by

$$\Delta f = (k\ f_{max})/n \tag{2-42}$$

where

Δf = the analysis bandwidth

k = the widening factor due to the weighting function (typically 1.5)

f_{max} = the upper frequency limit of the range

n = the number of analysis points

If greater resolution is required in a limited high frequency region, external frequency translators can be used to shift the desired region into the high-resolution range of the analyzer. For example, one can employ single sideband technique to scale the region, say, from 100 kHz to 100.5 kHz down to 0 to 500 Hz which can be handled with a real-time analyzer with 1.5 Hz resolution.

E. Other Techniques

1. Fast Fourier Transform

Recently, high-speed digital computation techniques have become available for spectral analysis. Foremost among these is the so-called Fast Fourier Transform (FFT) by Cooley and Tukey.[18] The FFT greatly reduces the number of steps required to perform a numerical Fourier analysis, which has made the calculation economically feasible for relatively large data samples. The advent of very high-speed minicomputers, especially those employing high-speed read only memories (ROM) for storing key portions of the FFT algorithms, has provided real-time FFT capability to beyond 20 kHz.

The key advantage of the FFT approach is that it provides phase as well as amplitude information on the signal components. Further, the FFT algorithms can be configured to calculate other analysis functions, correlation, convolution, cepstrum analysis, power spectrum analysis, etc.

In addition to the FFT, there are other techniques, such as the Walsh Transform which essentially performs a similar function using a different mathematical approach.

[18] J. W. Cooley and J. W. Tukey, "Algorithm of the Machine Calculations of Complex Fourier Series," Mathematics of Computation, Vol. 19, No. 90, April 1965, pp. 297-301.

2.4 TIME DEPENDENT FREQUENCY DOMAIN DATA REDUCTION

A. Introduction

In Section 2.3 the different categories of frequency domain data analysis equipment were described. These equipments are capable of analyzing a given time segment of data, producing a single plot of the average spectral levels versus frequency. When used in this manner the dimension of time is missing. With real life data the energy distribution is seldom stationary with time. Relative spectral energies shift in both amplitude and location as the scenario changes. Thus, as a rainstorm starts and builds up one might find an overall increase in broadband acoustic level. The song of a bird might cause the repetitive appearance of strong spectral lines. As a vehicle passes the data collection point, one would see spectral lines relating to the engine firing rate build up and decay. Their location in the frequency plane would shift depending upon the Doppler effect and the instantaneous firing rate. One can, of course, observe the changing pattern of data in an oscilloscope display of a real-time analyzer. This method, however, does not provide a hard copy of one spectral pattern for convenient analysis. Several means for reducing these data in time domain, by rapidly plotting or displaying these multiple "snapshot" spectra are available and will be described below.

B. Sonograph-Type Display

The basic problem involved in displaying time-dependent frequency-domain analyses is that of plotting three variables--level, frequency, and time--on a two-dimensional surface, i.e., paper, oscilloscope, etc. The different types of display equipment reflect different approaches to the solution of this problem.

An example of the sonograph-type display is shown in Figure 2-27. In this approach, the two axes represent frequency and time. The amplitude of the spectral component is represented by the density of darkness of the lines. Each new spectrum is displayed as a new line across the page (vertically).

The major limitation of a normal sonograph display is its relatively crude indication of spectral level. The dynamic range of the plot is such that a relatively few shades of gray can be distinguished. For this reason, an AGC is sometimes employed before the sonograph to adjust the signal level to its range of operation. The time and frequency definition of the sonograph can be quite good. Typically, 4 inches of paper is allocated for the frequency axis.

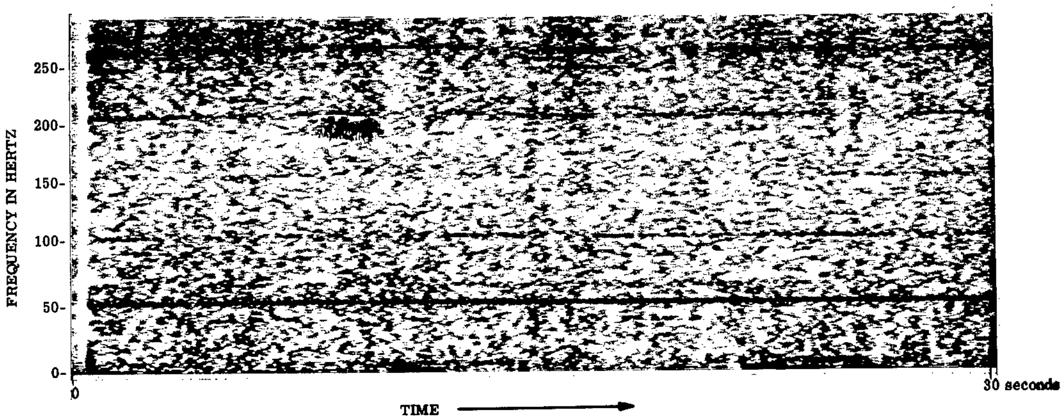


Figure 2-27. Example of Sonograph Display (Analysis Bandwidth 2.8 Hz)

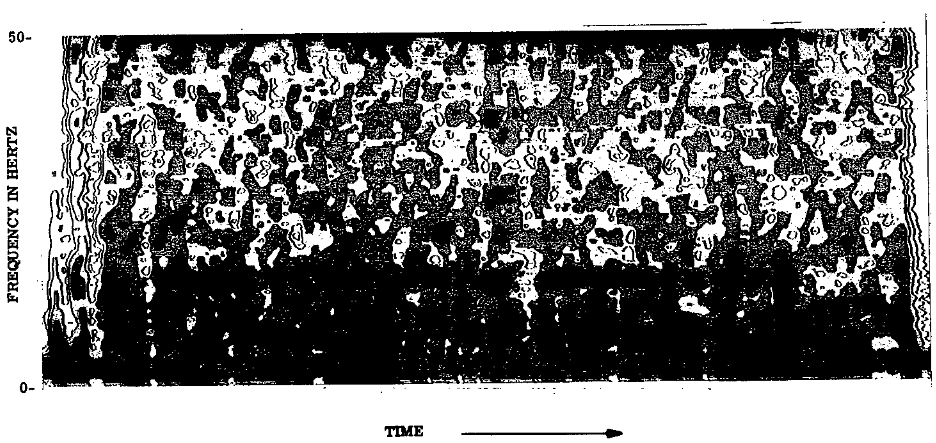


Figure 2-28. Example of Contour Graph Display

The major advantage of the sonograph is the ease with which one can identify subjects of interest whose signature consists of spectral lines whether stable in frequency or not. Such targets appear as a darkening of the trace at some point on the frequency axis. If the line is not frequency stable, the darkened section will take on a "slant" as a function of time. The "track" of the subject is still readily apparent.

C. Contour Graph Display

As with the sonograph, the contour graph displays frequency and time on the two major axes of the paper. However, rather than indicating spectral amplitude by intensity variations, the contour graph draws lines which are the locus of points of equal spectral energy. The result is analogous to a topographical map where lines of equal altitude are drawn--hence the name "contour graph." Typically a 42 dB dynamic range is accommodated utilizing seven shades of gray with a resolution of 6 dB for each shade.

As with the sonograph, the resolution of the frequency and time axes is quite good. However, the spectral level indication is necessarily discrete since a line is drawn through points of corresponding levels. The next line indicates a level X dB different from the previous one. One can interpolate between the lines to estimate spectral level but one cannot be confident of the distribution of levels between the contours. As the level difference between contours is diminished, the lines become somewhat difficult to follow.

On a contour graph subjects of interest having either line or narrow band signatures usually appear as a clustering of closed contours similar to the way a mountain, or if extended in time, a mountain range, would appear on a topographical map. An example of a contour graph display is shown in Figure 2-28.

D. 3-D Spectral Display

The 3-D spectral display is a device used with the real-time analyzers described in subsection 2.3.D. This approach attempts a pseudo three-dimensional display on the face of an oscilloscope by a technique similar to that used by a draftsman in making a projection drawing. The abscissa or X-axis represents frequency, the Y-axis represents time and the Z-axis amplitude. One spectra after another are plotted on the oscilloscope face with a slight shift in origin position to give the three-dimensional effect. The

approach is similar to the sonograph except that displacement in the vertical direction is used to indicate spectral level rather than density. This provides better resolution in spectral amplitude than the sonograph. The major disadvantage of the 3-D spectral display is the difficulty in obtaining a permanent copy since photographic means must be employed. Furthermore, although reasonably good resolution in the frequency and level axes are provided (limited by the size oscilloscope used) there is a limitation to the number of spectra, that is, the length of time which can be displayed simultaneously on the oscilloscope face. After the face is "full" the next display replaces the oldest one giving a history of the previous X seconds of data.

The presence of a line spectrum type signal on the display is indicated by a deflection at corresponding points on the frequency axis from spectra to spectra. Thus the "track" of the signal can be followed with time. An example of the 3-D spectral display is shown in Figure 2-29.

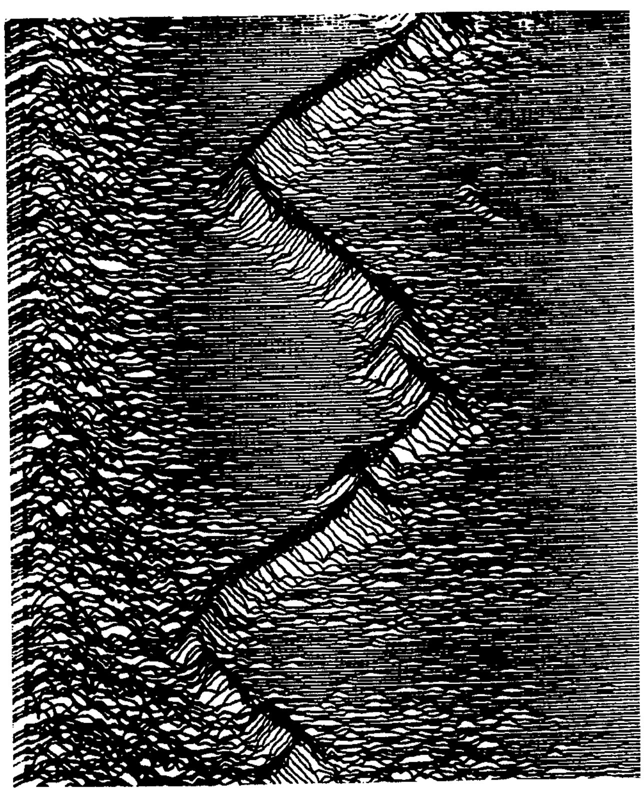


Figure 2-29 Example of 3-D Spectral Display

2.5 TIME DOMAIN DATA REDUCTION

A. Introduction

As discussed in subsections 2.2.F. through 2.2.H., it is useful to determine the characteristics of data in the time domain as well as, or in lieu of, the characteristics of the data in the frequency domain. An obvious example is the detection of an impulsive signal whose characteristics are much more readily identified by the shape of the pulse than by its frequency content. A more subtle example is the identification of, say, a certain type of vehicle whose distinguishing characteristic might be the rate of rise and the decay of sound level.

1. Raw Data and Envelope Data

In each of the above cases, a distinguishing feature would be found in time domain analysis. In the case of the impulse, the distinguishing feature was in the raw data; for example, the time rise of pressure pulse. In the case of the vehicle, the distinguishing feature was the rate of change of the average sound level, characterized by the slope of the rectified envelope of the sound pressure level. These two examples demonstrate two of the major categories of time domain data reduction, i.e., raw data reduction and display, and envelope or rectified data reduction and display.

2. Equipment for Time Function Display

The selection of a device or technique to display time functions is faciliated if the user is familiar with the limitations and special features of each of the techniques available. As of this writing, there are over 200 manufacturers of time function display equipment exclusive of those who produce oscilloscopes. The techniques which are currently being applied as well as a summary of their characteristics is shown in Table 1. There may well be improved systems or special models available which a particular manufacturer may have recently developed which may not be included in Table 2-1.

The selection of a display device should be approached by making a checklist of the available system parameters and their importance in a particular application. The following list represents the order in which the parameters are likely to be considered:

a. Frequency Response. Can the the recorder display the signal frequencies of interest, or the rate of change of the

Table 2-1. Equipment for Time Function Display

RECORDING TECHNIQUE	MAX. FREQUENCY LIMIT	REPRODUCIBILITY	ACCURACY % FULL DEFLECTION	TRACE WIDTH MAXIMUM RESOLUTION	FILM CAPACITY OR TIME FACTOR	INITIAL COST	OPERATING
Direct Writing Galvos							
A. Ink Pen	200 Hz	Excellent	1.0	0.05 mm	400' roll	100 - 3,000	Low - Med.
B. Heated Stylus	200 Hz	Excellent	1.0	0.05 mm	400' roll	100 - 3,000	Low - Med.
C. Pressure Sensitive	200 Hz	Excellent	2.0	Special paper 0.05 Smoked paper 0.005	400' roll Drum 24 Hrs	100 - 3,000	Spec. - Med. Smok. - Low
D. Electrosensitive	25 Hz	Good	2.0	0.05 mm	200' roll	5,000	Medium
E. Chopper Bar	1 Hz	Excellent	2.0	0.10 mm	200' roll	2,000	Low
Light Beam Techniques							
A. Conventional Photographic	13 kHz	Excellent	0.5	0.03 mm	400' roll	2,000 -10,000	High
B. Rapid-Develop	13 kHz	Fair	0.5	0.03 mm	400' roll	2,000 -10,000	High
C. Direct Printing Ultraviolet	13 kHz	Poor	0.5	0.03 mm	400' roll	2,000 -10,000	High
D. Direct Printing Electrostatic	300 Hz	Excellent	0.5	0.03 mm	400' roll	10,000	Medium
Potentiometer	1 Hz	Excellent	0.25	0.05 mm	200' roll	300-800/chan.	Medium
Sweep Balance	5000 Hz	Good		0.05 mm	100' roll	5,000	Medium
Ink Jet	1000 Hz	Good	2.0	0.01 mm	200' roll	3,000 -12,000	Medium
CRT Systems							
A. Conventional	1 MHz	Excellent	3.0	0.05 mm	Polaroid-single frame Developed-unknown	5,000	High

envelope? It is necessary to remember that an impulse display generally requires that the galvanometer have a natural resonance at least 1.6 times the highest frequency component of the pulse. This criteria is necessary to prevent the phase shift characteristics of the galvanometer from distorting the pulse. Other requirements follow from the rules for handling impulsive data outlined in subsection 2.2.D.4.

b. Resolution. Can the recorder resolve the high frequency components of the signal at a paper speed compatible with the total signal display duration? Systems with high resolution can operate at slower paper speeds for a given signal frequency. The total duration of the recording is often vital if the parameter being examined requires continuous or long duration display.

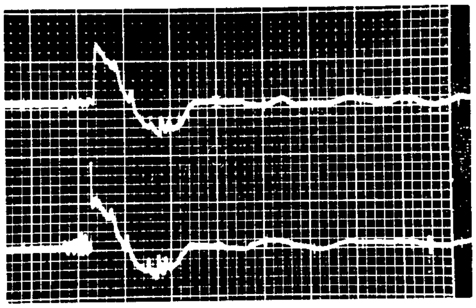
c. Recording Medium. The necessity for reproducibility and/or prolonged handling must be considered in selecting a technique. In addition, the cost of some recording mediums may put their use out of reach of a particular program. The selection of a medium is also affected by the length of rolls which are readily available if the display is of long duration. The selection of the recording medium is also closely associated with the necessary speed with which the finished record must be made available.

d. Portability. Will the recorder be used for laboratory use only, or will it be used for field display? The size, weight, and power consumption of recorders vary considerably and cover the entire range from a few pounds and with a clock spring drive all the way to hundreds of pounds with several hundreds of watts power consumption.

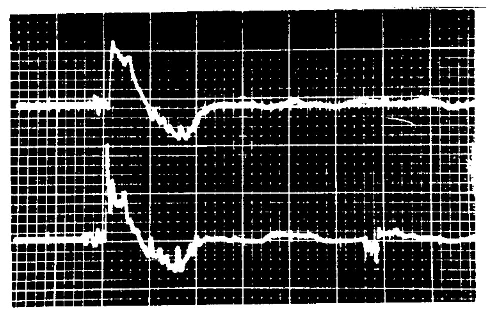
e. Compatibility. If currently there is display equipment in the data laboratory it is often important to remain compatible with the many expensive accessories which have been accumulated. This consideration is generally not important until the selection has been narrowed to a few systems, any of which could be used.

B. Raw Data Reduction and Display

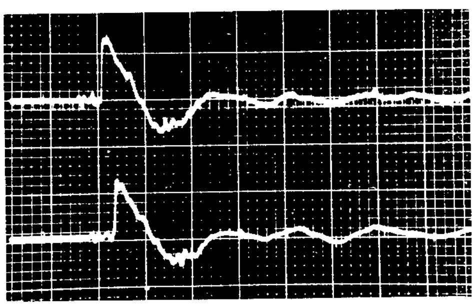
In general raw data is displayed when one wants to examine the instantaneous pressure of the disturbance. This happens most frequently when one is investigating impulsive phenomena such as explosions. In these cases the rise time of the impulse can be very short requiring a high speed display device, that is, one with extended frequency response such as an oscilloscope or light-beam oscillograph. An example of the raw data of sound pressure of explosions followed by reverberation and scattering is shown in Figure 2-30. When the raw data has relatively limited bandwidth pen recorders can be used for display.



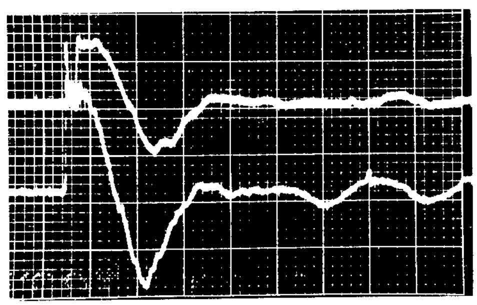
a. B&K 4134 at 0°, 50' From Source
Peak Pressure: 145.5 dB including spike
140.5 dB neglecting spike



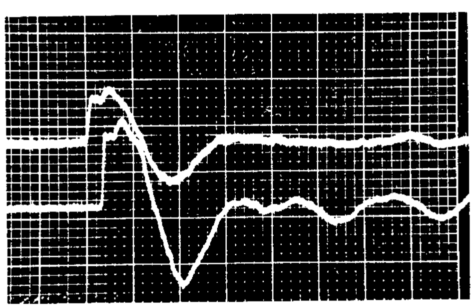
b. GR 1560-P5 at 0°, 50' From Source
Peak Pressure: 146.5 dB including spike
142.0 dB neglecting spike



c. GR 1560-P5 at 90°, 50' From Source
Peak Pressure: 142.0 dB



d. GR 1560-P5 at 0°, 140' From Source
Peak Pressure: 131.0 dB including spike
127.5 dB neglecting spike



e. GR 1560-P5 at 90°, 140' From Source
Peak Pressure: 126.0 dB

Figure 2-30. Typical Oscilloscope Traces of Measurements of Sound Pressure of Explosions Taken at Distances of 50' and 140' from the Event. Time-Base = 1 ms/cm; dB Reference = 0.0002 μbar

2.5.B-2.5.C

Frequently, to improve the signal-to-noise ratio of the display, the raw data is prefiltered to remove those frequency components not essential to the data. Care must be taken in selecting the prefilter. Generally, one performs a frequency analysis on the data first to determine the pass band of the major energy components and selects the filter accordingly. One must be cautious, however, to preserve the relative phase relationships between the components in the pass band. If a non-constant time delay (that is a non-linear phase) filter is used, the signal components will emerge from the filter at different times and the resultant signal will not resemble the original in either shape or peak level. Figure 2-31 compares the output from two low pass filters driven from a square wave source. The non-constant time delay of the Butterworth filter produces a ringing characteristic on the output signal which is not present on the Bessel filter output.

C. Envelope Data Reduction and Display

As described in subsection 2.2.H., envelope data reduction is most useful in establishing the appropriate AGC rates and in determining the best post-detection filter constants, for a given system. Based upon the rates of change of the average ambient levels and the rates of change of the desired signal level, one can select circuit constants to maximize the probability of detection.

1. Instrumentation for Rectified Data Display

Although one may employ many of the devices described in Table 1,it is frequently desirable to display the envelope of a given time sequence on a graphic level recorder. This device is commonly available and can produce a permanent, inexpensive, logarithmic level plot versus time. The relatively slow ballistics of the graphic level recorder can be modified with the aid of an electronic input processor to allow the recording of brief impulsive sounds, even if it cannot improve the resolution between rapidly repeated sounds. Such an electronic accessory also makes it possible to simulate the effect of meter or device ballistics. A system of this type has been employed in the preparation of the time domain information in several AWG reports and its description is included herein for this reason.

Figure 2-32 is a block diagram of the device. An input buffer amplifier is used to present a high impedance (100 K) to the data reduction recorder. The input buffer amplifier also provides a precise 12 dB gain and is capable of driving the 600 ohm decade attenuator which follows. A subsequent amplifier with

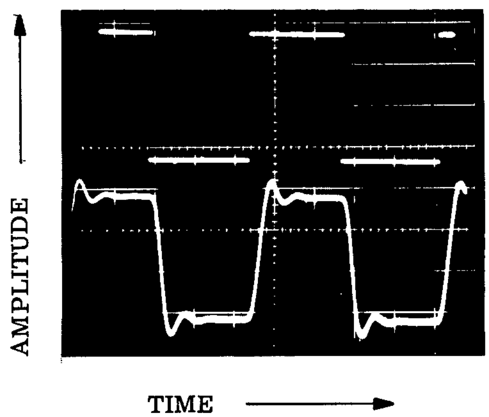


Figure 2-31a. Response of a 5-Pole Butterworth Low-Pass Filter to Square Wave

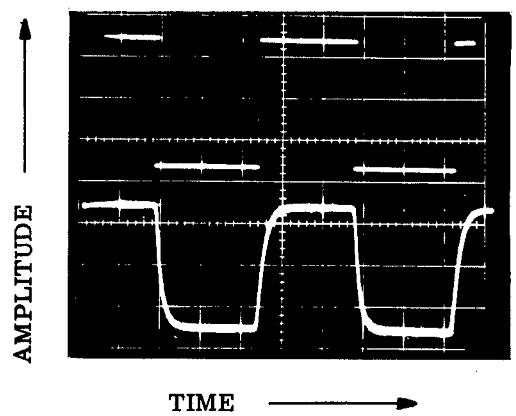


Figure 2-31b. Response of 5-Pole Bessel Low-Pass Filter to Square Wave

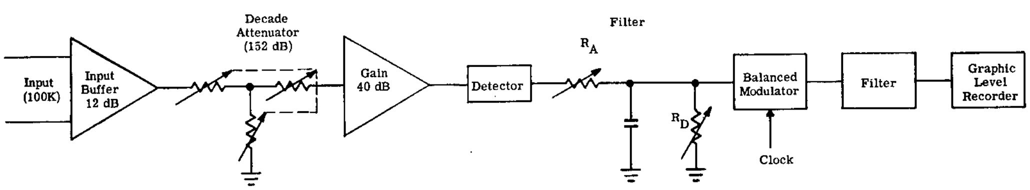


Figure 2-32. Block Diagram of Electronic Input Processor

2.5.C

40 dB gain terminates the decade attenuator and increases the signal level for the subsequent detector. This configuration allows the decade attenuator to be set for a 52 dB insertion loss. By reducing the amount of attenuation, the overall gain of the system can be controlled in a precise fashion. The detector can be either of the peak, averaging, or rms variety. For most of the AWG work, a wide dynamic range (in excess of 40 dB) full-wave rectifier was used as the detector. The detector is followed by an RC filter with variable attack and decay times. These time constants can be adjusted in accordance with the requirements of the data reduction. Typically, a decay time of 400 milliseconds is used to insure that the graphic level recorder pen reaches the maximum level of short duration signals. Depending upon the purposes of the reduction, attack times from 1 to 50 milliseconds have been used.

The logarithmic potentiometers which are typical of graphic level recorders function only with AC signals. Plotting time series data on a logarithmic scale is especially valuable to the circuit designer since growth and decay times of the time series can be read off directly in dB per second. Referring to the block diagram, a balanced modulator is used to convert the envelope level data to a proportional AC signal. A 10 kHz carrier frequency was used and the subsequent band pass filter was sufficiently wide to preserve all significant sideband components.

A plot of the linearity of the level analysis equipment described above is shown in Figure 2-33. The input level was varied in 5 dB steps, and the chart record shown a linearity within approximately 1/2 dB over a 40 dB range.

By properly adjusting the attack and release time constants, in conjunction with the damping of the pen motor servo-mechanism, the ballistics of a variety of devices can be simulated. As an example, Table 2-2 compares the chart-record results with readings from a standard sound-level meter. For steady-state signals, the two devices will indicate the same value. Over a fairly wide range of dynamic conditions, the chart record agrees within 1 dB in both maximum and minimum fluctuations with the readings of a sound-level meter.

Examples of time-domain analysis are shown in Figure 2-34.

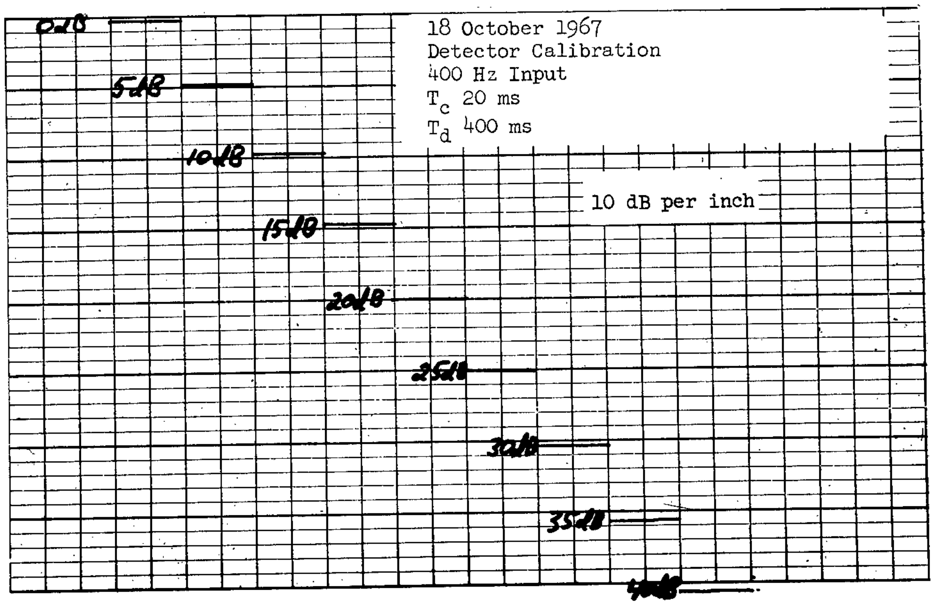


Figure 2-33. Detector Linearity Calibration

Table 2-2. Simulation of Sound Level Meter Readings on a Graphic Level Recorder

TIME ON (ms)	TIME OFF (ms)	SLM READING (dB)	BALLISTIC COMPENSATOR READING (dB)
31	127	9 ± 1/2	8-3/4 ± 1/2
31	63	7-1/2 ± 1/2	8 ± 1/2
31	31	6	6-1/2
31	15	3-1/2	4
31	7	1	3/4
31	3	-3	-4
63	127	7-1/2 ± 1-1/2	7-1/4 ± 1-1/2
63	63	5-1/2 ± 1/2	5-3/4 ± 1/2
63	31	3 ± 1/2	3-1/2 ± 1/2
63	15	0	1/4
63	7	-3	-4

2.6 SPECIAL DATA REDUCTION TECHNIQUES

A. Introduction

In addition to the standard frequency and time domain data reduction, other techniques are often employed to cull out special significant aspects of the data. For example, it might be significant to know the direction from which the sound arrived. In that case, one might employ directional microphones, or arrays of microphones, either directional or omnidirectional, from which the direction of arrival can be determined by special techniques. Or, one might find the amplitude variations of the data of special significance and so demodulate the signal before standard frequency and time domain reduction. Several of the more useful techniques will be discussed in this section.

B. Azimuth Location

One useful approach to determining the azimuth of sound arrival relies upon the special directional characteristics of the gradient microphone. This microphone, which in its simplest form consists of a diaphragm exposed to the sound on both sides, produces an output proportional to the pressure difference across the diaphragm, i.e., to the pressure spatial gradient.[19] (See Figure 2-35.) It can readily be shown that for wavelengths considerably greater than the microphone dimensions the output, E_1, is given by

$$E_1 = kp \cos \theta \qquad \text{volts} \qquad (2\text{-}43)$$

where:

- k is a constant determined by the microphone dimensions and the transducer sensitivity in volts per microbar
- p is the pressure of plane progressive sound wave in microbars
- θ is the angle between the direction of sound travel and the axis of maximum sensitivity of the microphone

If a second microphone, identical to the first is placed in close proximity but with its axis of maximum sensitivity oriented at right angles to the first, its output E_2 is:

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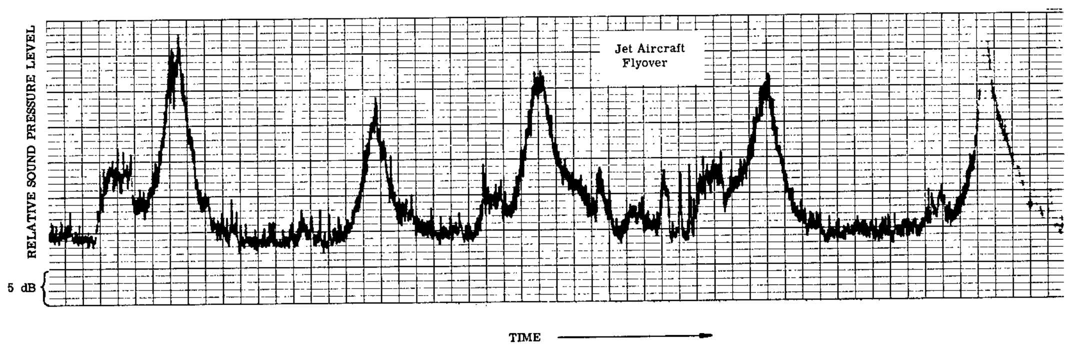
[19] See, for example, H. F. Olson, Acoustical Engineering, D. Van Nostrand Co., Inc., Princeton, N. J., 1957.



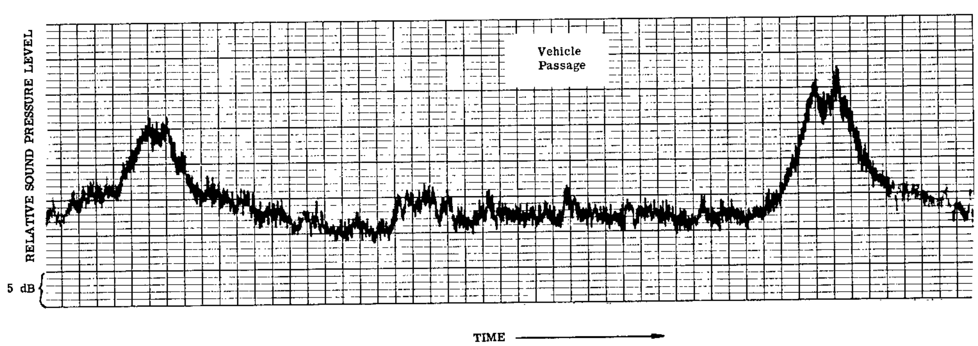
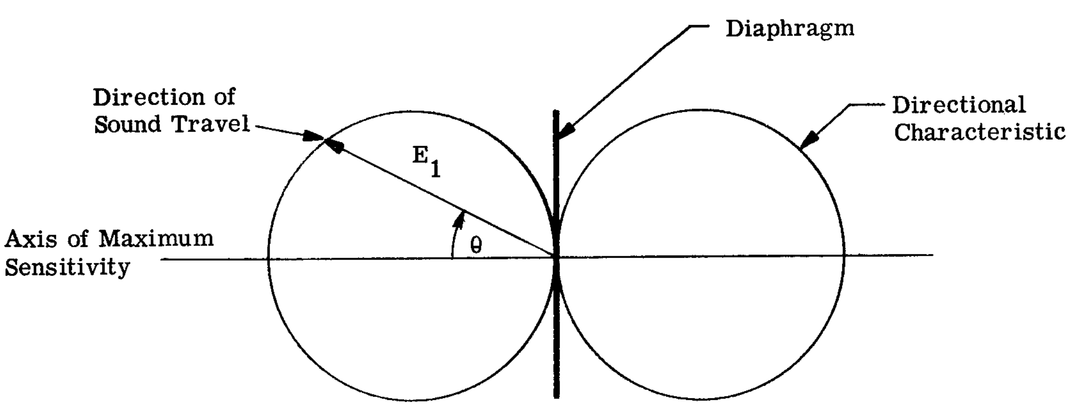


Figure 2-34. Examples of Time Domain Analysis



$$E_1 = kp \cos \theta$$

Figure 2-35. Directional Characteristic of Gradient Microphone

$$E_2 = kp \sin \theta \qquad \text{volts} \qquad (2\text{-}44)$$

where, k, p and θ are defined as before. Dividing the two signals gives

$$(E_2/E_1) = \frac{\sin \theta}{\cos \theta} = \tan \theta \qquad (2\text{-}45)$$

That is, the ratio of the two signals is equal to the tangent of the angle of sound arrival independent of signal strength. This approach has been used in the acoustic azimuth locator shown in Figure 2-36.[20] In this device the signal levels from a pair of matched gradient microphones after suitable filtering and processing are divided to display the azimuth vector to the sound source.

The same approach has been used very successfully in locating continuous sound sources such as vehicles. The line spectra from such devices allow one to select and lock on to a single line for analysis. This greatly improves the effective signal-to-noise ratio and thus the accuracy of the device. When tracking continuous line sources, a pressure microphone is normally incorporated in the array to eliminate the quadrant ambiguity in Eq. (2-45). The sense of polarity of the pressure signal [p in Eq. (2-43) and (2-44)] is given by this microphone so that the signs of E_1 and E_2 given by Eqs. (2-43) and (2-44) uniquely determine the quadrant from which the sound arrived.

C. Detection of Elevation

By combining several gradient and pressure microphones in various ways, a number of interesting functions can be obtained. For example, it becomes possible to determine the elevation of sound arrival as well as its azimuth. A microphone array utilizing these concepts is shown in Figure 2-37. By properly processing the outputs from a pressure microphone and a pair of space-orthogonally-related gradient microphones the microphone sensitivity patterns of Figure 2-38 can be achieved simultaneously from the one multiple sensor. Besides the omnidirectional pattern of the pressure microphone and the cosine patterns of the gradient microphones, a donut pattern (omnidirectional in the X-Y plane with first order gradient nulls in the vertical plane) and upward and downward oriented cardioid patterns are achieved. By different processing still other patterns can be obtained. By comparing the outputs from the different microphone subarrays as shown in Figure 2-39, the direction

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[20] ECOM-02226-F, "Development of 360° Acoustical Mortar Locator Experimental Development Model," Final Report, U. S. Army Electronics Command, Ft. Monmouth, N. J., March 1968

Figure 2-36a. Acoustic Azimuth Locator (Microphone Array)

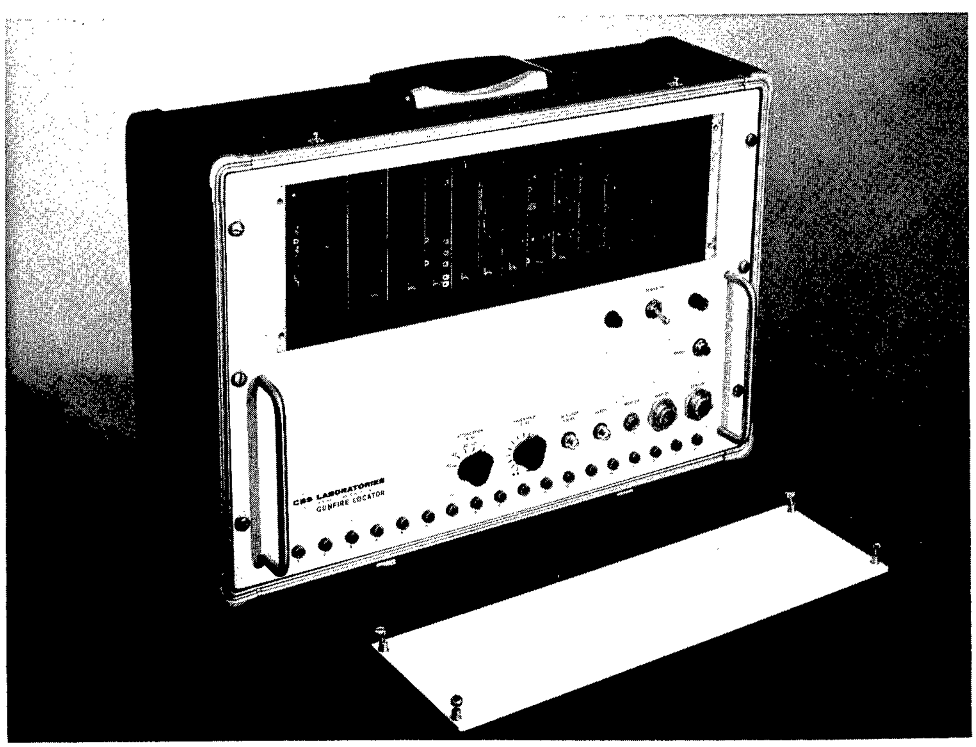


Figure 2-36b. Acoustic Azimuth Locator (Processor)

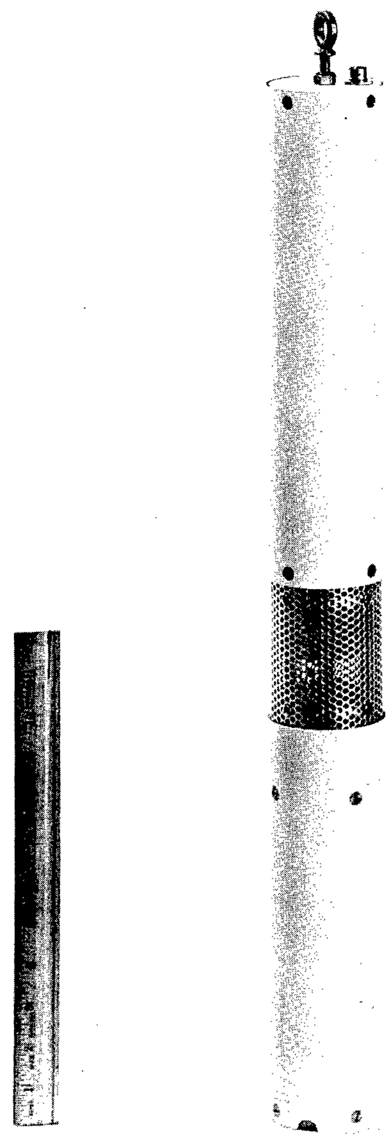


Figure 2-37. Apparatus for Simultaneous Generation of Multiple Microphone Patterns

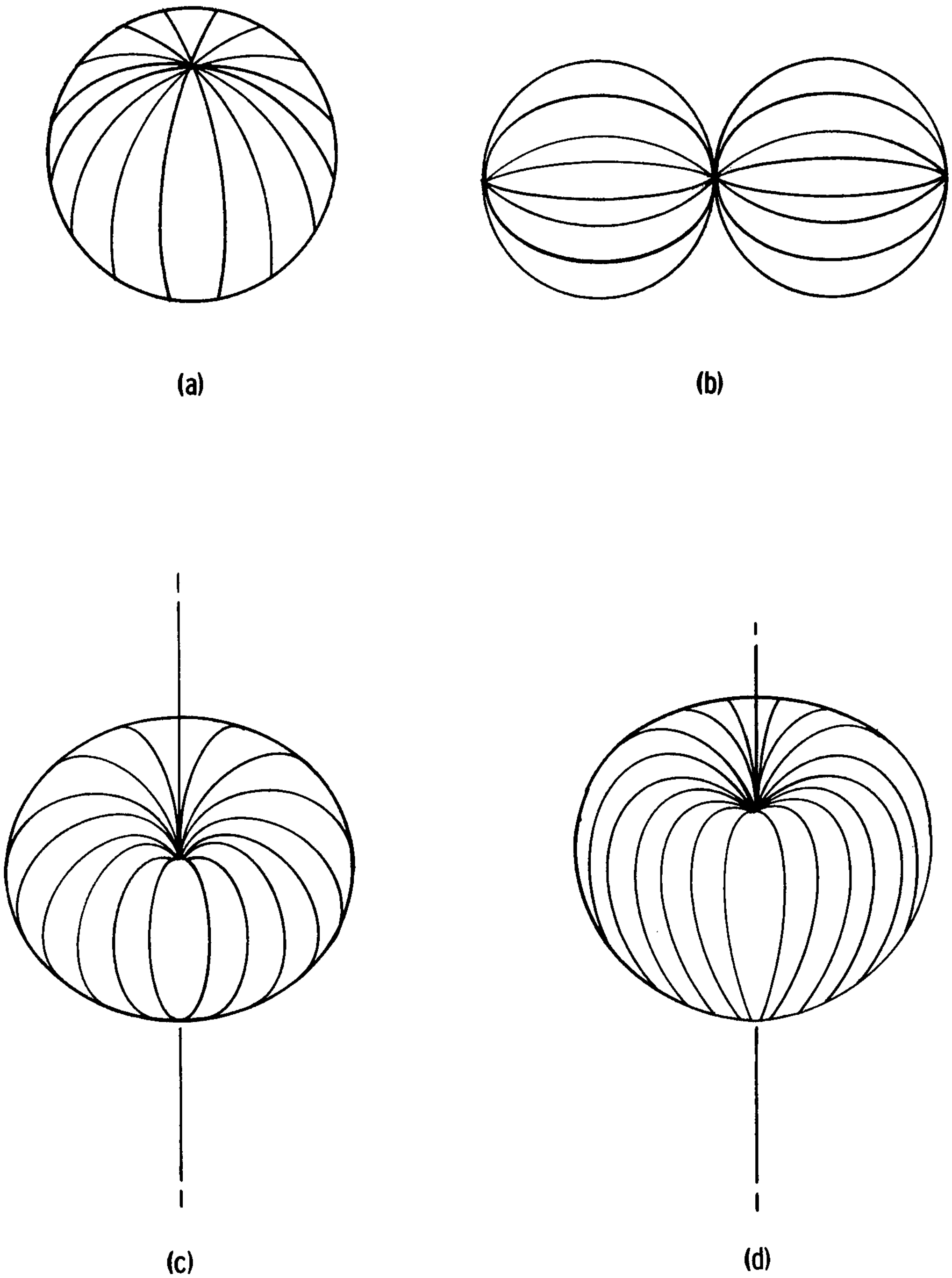


Figure 2-38. Microphone Sensitivity Patterns: (a) Omnidirectional Pattern; (b) Gradient Pattern; (c) Donut Pattern; (d) Cardioid Pattern

of sound arrival can be determined. In this case, all synthesized microphone patterns are normalized to unity for horizontally arriving sounds. The normalization applies to the maximum sensitivity point of the gradient transducers. It should be noted that in practical applications the microphone array should be located sufficiently above ground to avoid errors from ground reflections.

D. Detection via Correlation

A direction-finding technique which uses a completely different approach, but which also can determine the angle of arrival, uses cross-correlation between two spaced sensors. Unlike the comparative technique, the ability to determine more than one angle of arrival simultaneously also can be achieved. Basically, the correlation integral (see subsection 2.2.G) is,

$$G(\tau) = \int_{-\infty}^{\infty} g(t + \tau)\, f(t)dt \qquad (2\text{-}46)$$

where $g(t)$, $f(t)$ are the time domain signals from the two sensors, τ is a time displacement (between the functions) at which the correlation is computed. Suppose $g(t)$ and $f(t)$ are two outputs from the same source at different distances from the microphones. As τ is varied continuously in a cyclical manner, each time it corresponds with the delay caused by the difference in path length between source and sensors, a maximum in $G(\tau)$ will occur. If the sensor spacing is small compared to the distance to the source, it becomes simple to convert τ directly into angle. Zero τ denotes a source always on a line that is the perpendicular bisector of the line connecting the two sensors. If the source is continuous, one can also effect a correlation. However, in a plot of $G(\tau)$ more than one peak can occur. Figure 2-40 is a plot of $G(\tau)$ along the abscissa for a truck moving approximately along the perpendicular bisector of the line connecting the microphones, since the correlation peaks around zero degrees.

Figure 2-41 is a plot of the correlation function between a pair of vertically displaced microphones with an aircraft as the sound source. In this case, besides the peak correlation achieved initially at about zero degrees, there is evidence of secondary correlation peaks due to sound reflections. Figure 2-42 is similar to Figure 2-41 except that only the correlation peaks are plotted. This "enhancement" makes the data easier to interpret.

Although correlation has advantages over the comparative

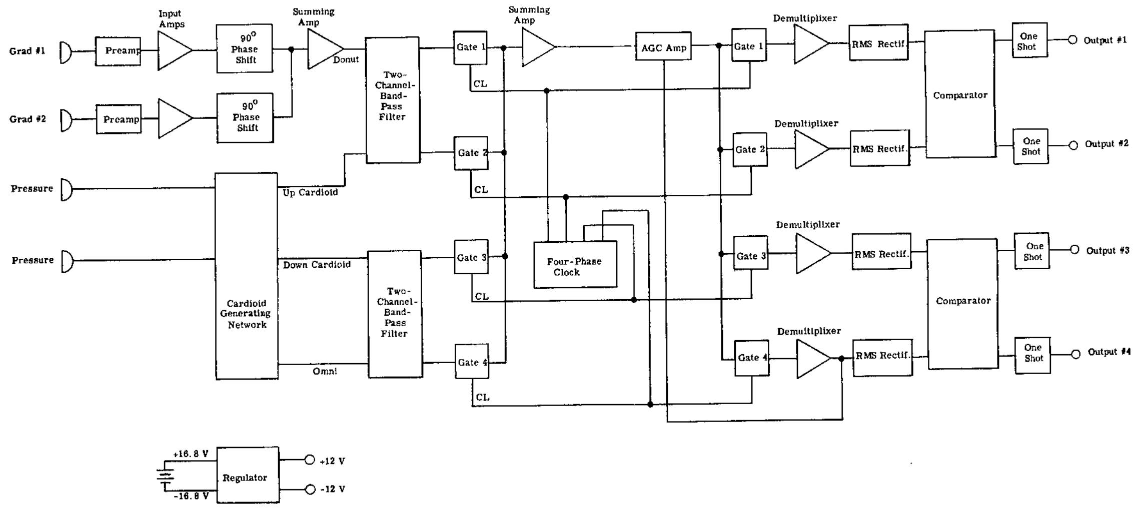


Figure 2-39. **Block Diagram** of Up/Down Sensor

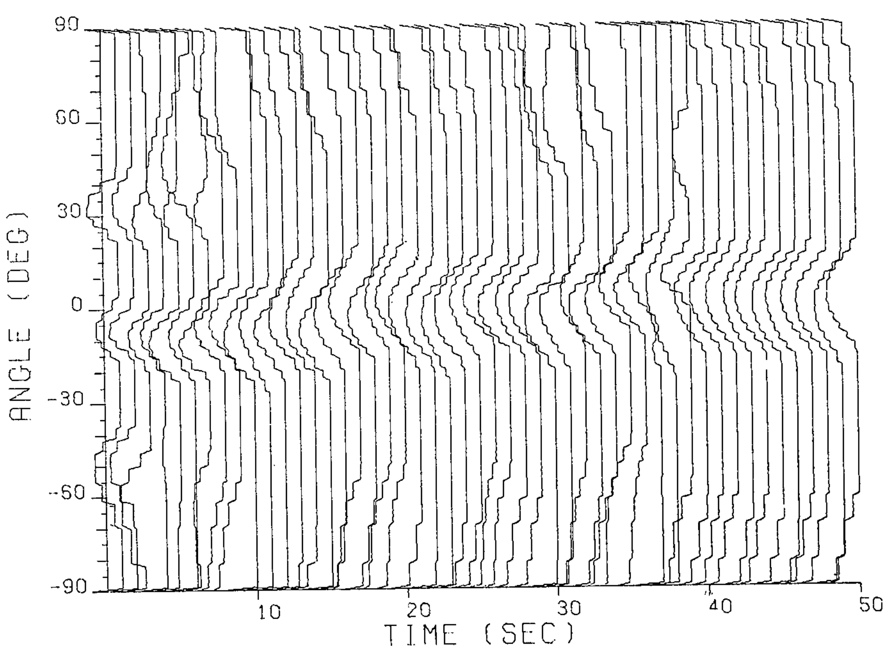


Figure 2-40. Correlation Plot of Moving Vehicle

system (multi-angle determination and no input calibration necessary), it does suffer in requiring reasonable physical dimensions for a given frequency; the distance required is normally many wavelengths; without this the correlation peak would be very small. Another problem occurs if the source produces a single frequency. Then the correlation maximum is not unique and will occur at every τ equal to the period of the signal. In general, real life targets with cyclical outputs such as aircraft have sufficient non-redundancy in correlation peaks to produce a maximum.

E. Arrays

To cover the subject of steerable arrays in depth is beyond the scope of this chapter. However, it is useful to discuss briefly a simple array to illustrate the concept and indicate the power of the approach. Arrays are constructed from multiple microphones spaced apart from each other. The outputs from the microphones are combined. In the simplest case the microphones are omnidirectional or pressure units although directional sensors can be employed to advantage. Typically the outputs are summed usually with different weighting factors and possibly with different time delay networks. The result is a polar sensitivity pattern which is highly directional, albeit, the directionality is a function of frequency or, more precisely, of wavelength. The array theory is analogous to that employed in the design of radio antenna arrays and is similar in many respects to diffraction in optics. Consider an array of four microphones, 1, 2, 3, and 4, equally spaced with intersensor distance ℓ (Figure 2-43). Assume the outputs are added so that

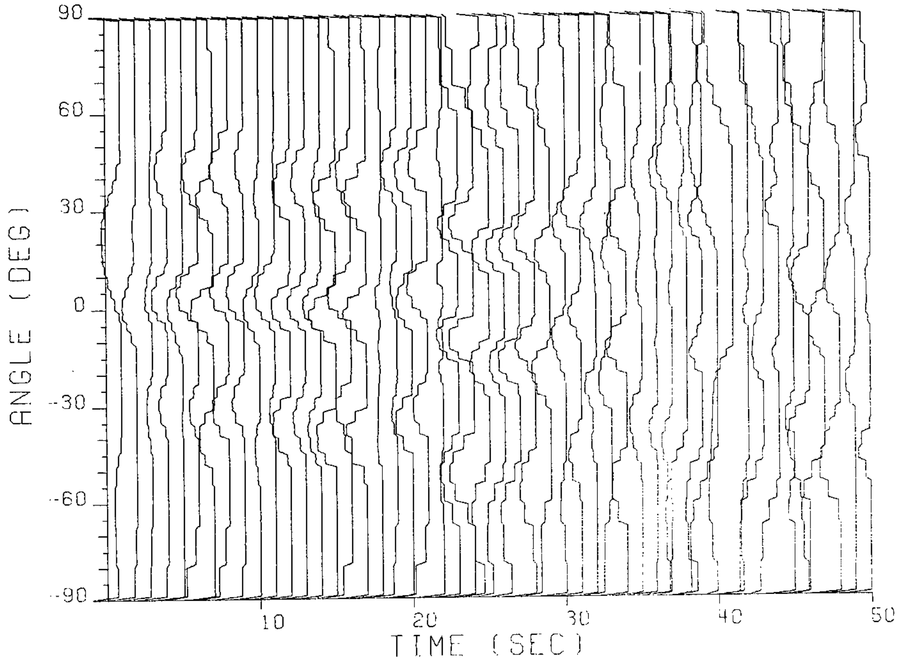
$$E = E_1 + E_2 + E_3 + E_4 \tag{2-47}$$

where the E's are the instantaneous values of voltage output of the respective microphones.

Assume a sinusoidal sound source at infinity so that the wave front is parallel and of equal peak pressure p_m at each sensor. Then for a wavefront at angle θ to the array,

$$E_1 = kp_m \exp j(\omega t + \frac{3}{2}\frac{\ell\omega}{c}\sin\theta) \tag{2-48}$$

$$E_2 = kp_m \exp j(\omega t + \frac{1}{2}\frac{\ell\omega}{c}\sin\theta) \tag{2-49}$$



AIRCRAFT - TAPE 1 - BEGIN 100300 - SEP=9.5 FT - VERTICAL - SRATE=2000

Figure 2-41. Correlation Plot from Aircraft

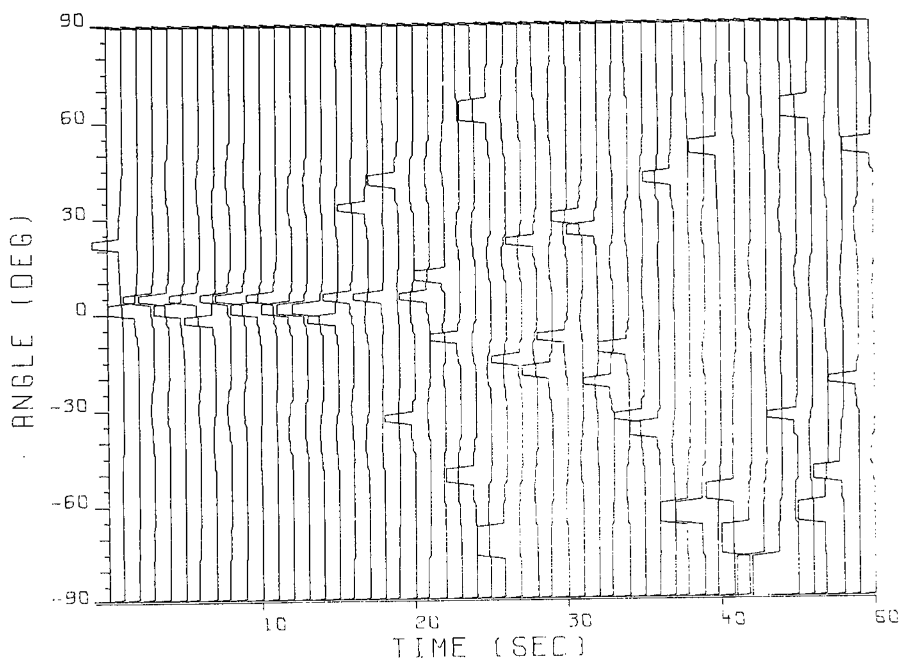


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Figure 2-42. Enhanced Correlation Plot from Aircraft

2.6.E

$$E_3 = kp_m \exp j(\omega t - \frac{1}{2}\frac{\ell\omega}{c}\sin\theta) \tag{2-50}$$

$$E_4 = kp_m \exp j(\omega t - \frac{3}{2}\frac{\ell\omega}{c}\sin\theta) \tag{2-51}$$

where:

k is the microphone sensitivity constant

ω is the angular frequency of the source

c is the velocity of propagation

By defining

$$X = \frac{\ell\omega \sin\theta}{c} = 2\pi \frac{\ell}{\lambda}\sin\theta \tag{2-52}$$

and by normalizing to

$$kp_m = 1 \tag{2-53}$$

Equation 2-47 becomes

$$E = \exp j(\omega t)\ \{[\exp j(\frac{3}{2}X) + \exp -j(\frac{3}{2}X)] + [\exp j(\frac{1}{2}X) + \exp -j(\frac{1}{2}X)]\} \tag{2-54}$$

$$= \exp j(\omega t)\ [2\cos(\frac{3}{2}X) + 2\cos(\frac{1}{2}X)] \tag{2-55}$$

For spacing of 1/4 wavelength

$$X = \pi/2 \sin\theta \tag{2-56}$$

so that the normalized output in dB as a function of θ for $\ell = \lambda/4$ is:

$$\text{Output} = 20 \log \{2[\cos(\frac{3\pi}{4}\sin\theta) + \cos(\frac{\pi}{4}\sin\theta)]\}, \text{dB} \tag{2-57}$$

which is plotted in Figure 2-44. The gain for $\ell = \lambda/2$ is plotted in Figure 2-45.

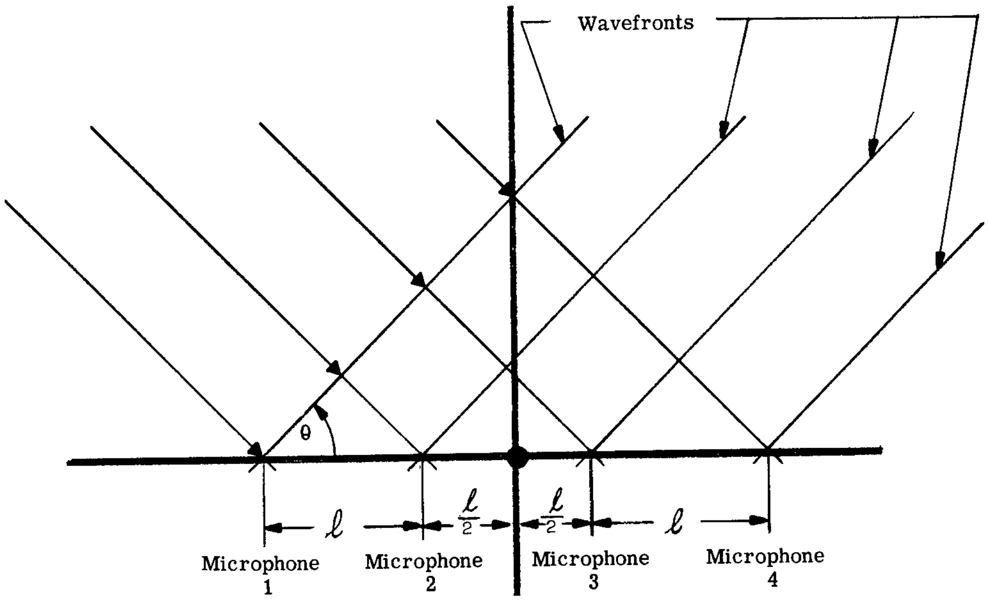

Figure 2-43. Four-Microphone Linear Array

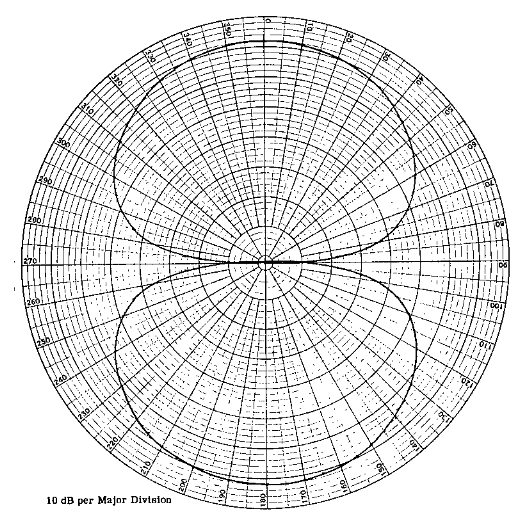

Figure 2-44. Four-Microphone Linear Array Polar Pattern for $\ell = \lambda/4$

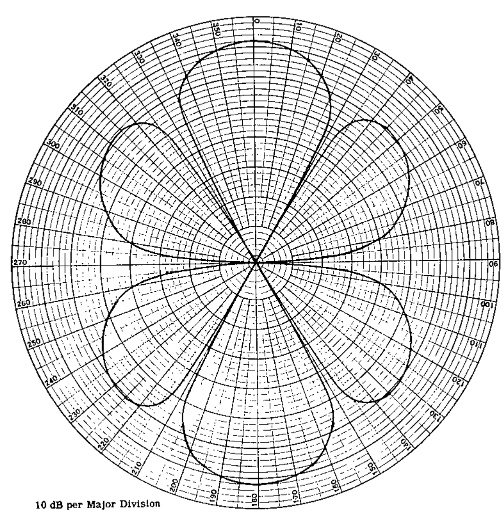

Figure 2-45. Four-Microphone Linear Array Polar Pattern for $\ell = \lambda/2$

2.6.E-2.6.F

The above exercise illustrates the patterns which can be achieved by linear arrays and shows that the pattern is frequency sensitive. By using more microphones, higher order patterns can be achieved. By combining the signals through controllable delays, the pattern can be electrically steered.

F. DEMON

Occasionally, the rate of variation of the signal strength is of more significance than the signal itself. For example, the "beating" between two engines may be more characteristic of a device than the engine sound itself. In such a case the base signal can be considered as a carrier which is amplitude modulated by the desired information. Then it is appropriate to demodulate the signal before analysis. This procedure goes under the appellation, DEMON.

One ordinarily thinks of the amplitude modulation process as one that involves a carrier of given frequency modulated in amplitude by information whose components have frequencies that are much lower than the carrier frequency. For the simple case of a single frequency as the modulation, the resultant waveform can be expressed as:

$$e(t) = A[1 + m \cos(\omega_m t + \phi_m)] [\sin(\omega_c t + \phi_c)] \qquad (2\text{-}58)$$

where

$e(t)$	=	instantaneous amplitude of the waveform	(volts)
A	=	the amplitude of the carrier	(volts)
m	=	the modulation index	(numeric)
ω_m	=	the modulating angular frequency	(radians/sec)
ω_c	=	the carrier angular frequency	(radians/sec)
ϕ_m	=	the phase of the modulating signal	(radians)
ϕ_c	=	the phase of the carrier signal	(radians)

To obtain the information, $mA \cos \omega_m t$ from the composite amplitude modulated signal, one feeds the signal through an AM detec-

tor (or demodulator) which strips off signal amplitude and disregards the carrier. One way of looking at the demodulator is to consider it an item that is sensitive only to amplitude variations and not to phenomena such as frequency or phase variations which are in quadrature with amplitude variations.[21] This is to say that if the carrier were to vary in frequency over some given range, the demodulated output would not be affected.

The above concept of amplitude modulation may be broadened by allowing the carrier frequency, amplitude and phase to change. This results in a waveform

$$e_2(t) = [f(t)]\,[1 + m(t)\,g(t)] \qquad (2\text{-}59)$$

where

$e_2(t)$	=	instantaneous amplitude of the waveform	(volts)
$f(t)$	=	instantaneous amplitude of a random waveform of given bandwidth	(volts)
$m(t)$	=	modulation index	(numeric)
$g(t)$	=	instantaneous amplitude of a waveform having frequencies much lower than those in $f(t)$	(volts)

When $e_2(t)$ is passed through an AM detector, the output is essentially $g(t)$, that is, the output is the envelope of $f(t)$.

A sound analog of $e_2(t)$ may be encountered. It may be seen that there is envelope information in the composite signal so that AM detection must first be effected to reduce this type of data.

The usefulness of the DEMON technique is demonstrated in Figures 2-46a and 2-46b which depict identical frequency analyses of the same data, one before and one after demodulation. A distinct structure is apparent in the demodulated data which does not appear in the analysis of the raw data.

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21 See A. Hund, Frequency Modulation, McGraw-Hill Book Co., Inc., 1942, p. 3.

2.7 MODELING

A. Introduction

The term model, as used in signal analysis, is a mathematical expression describing a phenomenon in general terms. The purpose of the model is to provide a means for estimating the performance of a system or a device mathematically without physically testing it. Thus, a design may be evaluated even before it is engineered or produced, or a complete scenario can be analyzed to predict a system performance. Obviously, the validity of the evaluation depends upon the validity and completeness of the model. Modeling for acoustic detection systems encompasses not only the system model, but source, noise, and propagation models. These techniques are discussed in this section.

B. Basic Considerations for Modeling

The function of a model is to provide a simplified mathematical expression approximating the behavior of a real function to an accuracy sufficient for the purpose for which the model is to be used. Models should have several important characteristics.

1. Their accuracy should be commensurate with the accuracy of the data. Typically, this means that the model should fit the data to within a few decibels. Accuracies greater than a few decibels usually are not warranted because of the accumulation of inaccuracies in microphone calibration, distance estimation, recording accuracy, analysis equipment accuracy and linearity and sensitivity of the simulated system to small changes in signal level.

2. They should include all parameters of significance. Significance is a quantitative term rather than a qualitative term, and a parameter of significance is one which affects the model by more than a few decibels.

3. They should be of a form suitable for entry into a computer. This means that they should be piecewise linear in decibels, and without significant discontinuities. If this is not possible, they should be given in tabular form.

4. In view of the general inadequacy of theories based on excessively idealized models, the models should conform to measured parameters rather than calculated ones. Only in the event that some important parameter cannot be measured should theoretical results be used. In every case where measurements conflict with theory, the results of measurements should be employed.

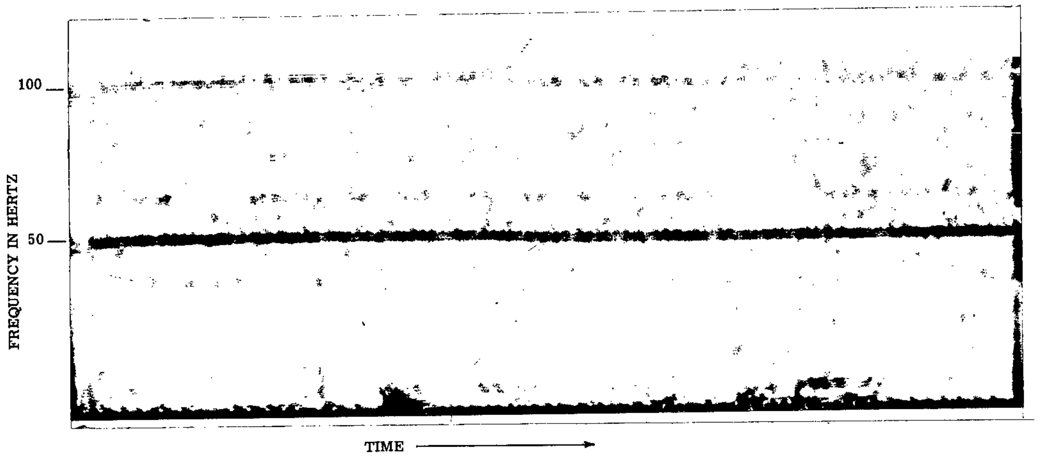

Figure 2-46a. Sonograph Display of Raw Data

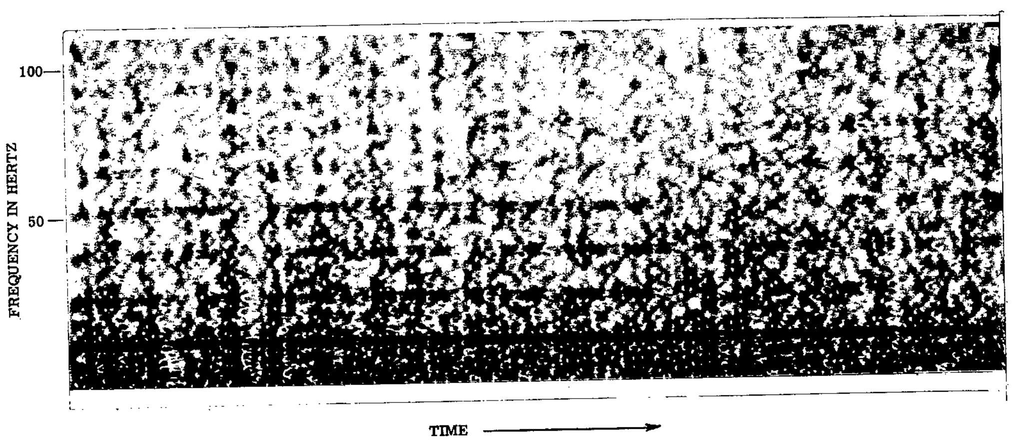

Figure 2-46b. Sonograph Display of DEMON Data

C. Model Structure, Sources

For a continous spectrum, the model should be a piecewise continuous approximation to the measured spectrum. The preferred approximations are constant, linear and logarithmic in decibels with frequency in that order. The continuous structure will be a mean spectrum averaged over as many spectra as are stable.

For a line spectrum, the model should contain all significant lines. The line strength in SPL should be given for each line. The SPL values should be the mean averaged over as many spectra as are stable. If possible, a simple equation should be given relating the frequency and level of each of the lines. Where this is not possible, the model, based on measurements, should be given in the form of a table.

Where some parameter of the model, such as SPL or line frequency is observed to be a function of time, the time-dependent function should be expressed in the simplest reasonably appropriate terms. If the variation is periodic or nearly so, then the time function should be expressed as a periodic function having parameters approximating the mean of those observed. If the variation is random, distributions and variances approximating the mean observed characteristics should be used.

Sound propagating from sources moving in space often may be considered to have the appropriate theoretical Doppler shift, but otherwise to have infinite velocity of propagation. In other words, the sound may be considered to arrive at the sensor in the same instant that it departs the source. When arrays of microphones or large-scale scenarios are used, propagation velocity becomes important; its effect should be considered in the model.

For those sources which may be localized in space, such as aircraft and vehicles, SPL and PSL values are usually given, referred to their value one meter from an equivalent point source. For those sources which cannot be localized in space, such as environmental background noise, it is usually convenient to refer to the SPL and PSL values at the microphone.

D. Model Structure, Transducers

The directional sensitivity of the transducers is customarily given in three-dimensional space in either Cartesian or spherical coordinates.

For each transducer, the ratio between the rms voltage that would be produced in a perfectly diffused sound field of constant frequency and given sound pressure and the rms voltage produced in response to a plane progressive sinusoidal sound wave arriving in the direction of maximum sensitivity should be given. This ratio is termed the Random Response.

The formulae for directional sensitivity of a number of transducers can be found in standard acoustical texts.[22]

E. Model of Sound Propagation

The process of sound traveling through a complex propagation environment may be mathematically modeled in terms of environment parameters. A first-order modeling technique treats the environment as an invariant linear filter, and the sound sources as a time waveform (signature) whose frequency components are determinable. Both the source-signature and the filter characteristics of the environment are determined experimentally. The mathematical model then becomes:

$$SPL_r(f, r, \theta) = SPL_o(f, \theta) + B(f, r, \theta) \qquad (2\text{-}60)$$

(dB re 0.0002 μbar)

where

$SPL_r(f, r, \theta)$ = sound pressure level of a given frequency component of the signature at range r, and angle θ from the source in dB re 0.0002 μbar

$SPL_o(f, \theta)$ = sound pressure level of a given frequency component of the signature, at unit distance from the source, in a given direction θ in dB re 0.0002 μbar.

$B(f, r, \theta)$ = the attenuation, dB, caused by the environment in a given direction, at a given frequency and as a function of range

r = range from source

[22] See footnote 19, p. 2-78

2.7.E

Eq. (2-60) may be modified to give the following approximate form:

$$SPL_r(f, r, \theta) = SPL_o(f, \theta) - 20 \log r - \alpha(f)r \qquad (2\text{-}61)$$

(dB re 0.0002 μbar)

$\alpha(f)$ = the attenuation coefficient at frequency f due to the propagation characteristic of the environment, dB/m or dB/ft

For the Eq. (2-61) spherical sound spreading and a constant attenuation with range, due to the media, are postulated.

The terms $SPL_o(f, \theta)$ and $\alpha(f)$, in Eq. (2-61) may be determined by measuring sound pressure levels of each different frequency at different ranges within the propagation environment. The recording ranges chosen should not be in the near field of the source; this means that the signatures are recorded at a range several wavelengths greater than the wavelength of the lowest frequency present. A convenient source of energy for different frequencies is an impulsive sound source like an explosion of a small charge of TNT or a pistol shot. To illustrate how SPL_o and $\alpha(f)$ are determined consider that an explosion is set off and signatures at ranges 100, 200 and 400 m from the source are recorded. Also, suppose that when these signatures are reduced, using techniques in Section 2.3, the following data for one frequency band is obtained, and the set of calculations shown below is performed:

Range r, meters	100	200	400
SPL_r (dB re 0.0002 μbar)	25	17	8
Spreading loss correction, 20 log r	40	46	52
$SPL_o - \alpha r$	65	63	60

The results are plotted as shown in Figure 2-47.

The slope of a regression line through the points is $\alpha(f)$, in dB/m. The dispersion of the measured points about the line indicates how well the constant factor $\alpha(f)$ models the attenuation phenomenon. The intersection of the line with the ordinate at $r = 0$ is $SPL_0(f, \theta)$ for the given environment.

Equation (2-61) for the above conditions (which we designate A) may then be rewritten as follows:

$$SPL_r(f_A, \theta_A) = 66.5 - 20 \log r - 0.0162r \quad \text{(dB re 0.0002 µbar)} \quad (2\text{-}62)$$

The above procedure may then be carried out for remaining pertinent frequencies and directions of propagation that characterize the source and the medium. Results may then be extrapolated to fit other similar environments, or other ranges not measured; care is necessary not to overextend the extrapolation. Normally, the experiment would be done many times with many more data points. When a line is fitted to the data, the scatter will give an idea of the uncertainty of the data.

A more sophisticated approach to modeling, yielding approximately the same result but a better statistical determination of the parameters involved, is the use of regression analysis. Here, more complicated forms of modeling can be investigated with comparative ease, along with an ability to determine the significance and confidence of the various terms.[23]

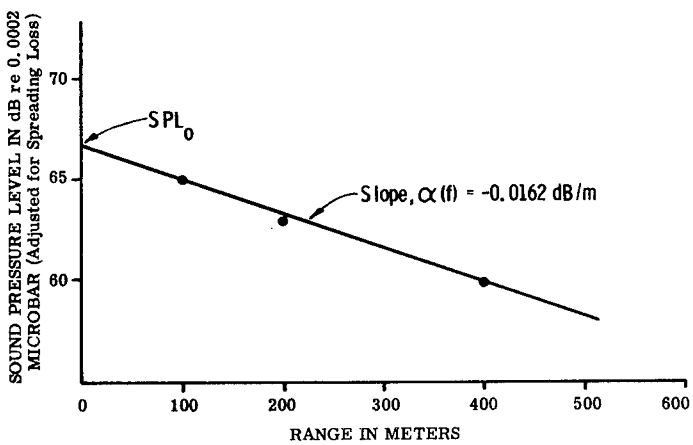

Figure 2-47. Graph for Estimating SPL_0 and α

[23] J. R. Harris, "A Study of Using Multiple Nonlinear Regression Analysis for Acoustical Modeling," Report No. NADC-SD-7027, Naval Air Development Center, Warminster, Pa., 19 June 1970.

CHAPTER THREE

PROPAGATION

CHAPTER THREE: PROPAGATION

3.1 INTRODUCTION

Sound waves travel from source to receiver through open atmosphere that is by its nature in constant motion and fluctuation. Density and temperature, wind and humidity, are never uniform in a given volume of open air under observation nor are they constant in time. The longer the transmission path through atmosphere, the more important is the effect of the interposed ground conditions on the received sound. Because the atmosphere is neither homogeneous nor quiescent and the ground covering can vary to a large extent, propagation of sound in open space constitutes a complicated statistical problem. It is the purpose of this chapter to present a combination of theory and empiricism on propagation studies made of acoustical signals in various types of atmospheric conditions and foliage using as a basis the experimental data gathered in the years between 1966 and 1970.

In the study of propagation of sound, two approaches usually are used, one being the ray path theory and the other the normal mode theory. Each of these approaches is best adapted to specific situations. The ray path method is appropriate to the study of propagation in the atmosphere when the boundary conditions are mainly due to ground. This approach is treated in subsections 3.3.A through 3.3.D. The normal mode is more convenient when multiple and well-defined boundary conditions exist such as those we find in shallow water with both the bottom and surface reflections present. This method is discussed in Section 3.4. The real world problem in air is complicated by scatter as described in Section 3.5. Empirical fit is appropriate in these cases with the modeling being determined by physical considerations as derived from ray path theory, which is discussed in Section 3.2. In the studies undertaken, except for a few isolated cases where strong temperature inversions and/or heavy canopy foliage allowed some channeling to take place, this latter type of modeling appeared to be quite adequate. Other aspects of modeling are treated in Section 2.7.

3.2 ANALYSIS OF SOUND PROPAGATION

A. Introduction

Ordinarily, the decrease in sound-pressure level at a distance from the source in an ideal, homogeneous, loss-free atmosphere is most affected by the spreading of the sound waves which are radiated by the source. The sound-pressure amplitude from a point source varies inversely as the distance from the source. That is, there is a drop in sound-pressure level of 6 dB with each doubling of distance away from the source.

At distances large compared with its size, the sound source can be treated as a point, even though it is not physically very small. If we know that sound-pressure level L_o at distance r_o, theoretically the sound pressure level L in the same direction at another distance r is given by

$$L = L_o - 20 \log (r/r_o), \text{ dB} \qquad (3\text{-}1)$$

B. Excess Attenuation

It has been established experimentally that the level of the received signal after it has travelled some distance along the ground is almost always below the level which would be expected if sound propagation had taken place in homogeneous air at rest over a perfectly smooth reflecting ground plane. The difference between the attenuation actually observed and that due to spherical spreading alone is frequently referred to as "excess attenuation" (A_e).[24] This is a useful quality for describing the transmission path between source and receiver acoustically. The excess attenuation depends, generally speaking, on the frequency of the transmitted sound and the distance between source and receiver and on the ever changing characteristics of the atmosphere. The sound pressure level equation is thus modified as follows:

$$L = L_1 - 20 \log r - A_e, \text{ dB} \qquad (3\text{-}2)$$

The excess attenuation A_e depends on the following factors:

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[24] L. L. Beranek, ed., Noise and Vibration Control, McGraw-Hill Book Co., Inc., New York, 1971, p. 169. It is equivalent to the quantity "transmission anomaly" in Principles and Applications of Underwater Sound, U. S. Gov't. Printing Office, NAVMAT, P-9674, Dept. of the Navy, 1968, p. 8.

1. molecular absorption of sound in the air

2. refraction of sound waves due to the presence of vertical temperature gradient and wind gradient

3. attenuation and scattering of sound by trees, bushes, and other solid obstacles

The contributions of each of the above factors to the overall excess attenuation are not strictly independent in the general case although, as a first approximation, some of these effects can be regarded as independent. Under most practical conditions, the excess attenuation at a given frequency of the sound is directly proportional to distance. So we have

$$L = L_1 - 20 \log r - \alpha r, \text{ dB} \qquad (3\text{-}3)$$

where α designates the excess attenuation per unit distance. Eq. (3-3) of course expresses the same law as has been previously formulated in Eq. (2-61) assuming that the direction of propagation θ is a constant.

C. Dependence on Humidity and Scattering

The molecular absorption of sound energy in air depends on frequency and humidity. It has been determined theoretically[25] and confirmed experimentally[26] by a number of investigators. Over open level ground, appreciable vertical temperature and wind gradients almost always exist; the former because of the heat exchange between the ground and the atmosphere, the latter because of the friction between the moving air and the ground. Because of these gradients, the speed of sound varies with height above the ground and sound waves are refracted, that is to say, bent upward or downward. Under these conditions, the straight line transmission characteristic of the sound is reduced.

Scattering refers to deflection of sound energy from its

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[25] H. O. Kneser, "Interpretation of the Anomalous Sound Absorption in Air and Oxygen in Terms of Molecular Collisions," J. Acoust. Soc. Am., Vol. 5, 1933, p. 122.

[26] E. J. Evans, and E. N. Bazley, "The Absorption of Sound in Air at Audio Frequencies," Acoustica, Vol. 6, 1956, p. 238.

original path by obstacles suspended in air; thus being diminished in the direction along which the receiver is located. Sound traveling through dense woods and shrubbery is attenuated through absorption by leaves on trees and on the ground and by multiple scattering by the tree trunks and limbs.[27] No comprehensive theory exists concerning the jungle acoustics at this time. Some results of experimental efforts have been published[28,29,30] which this report will attempt to interpolate and extend in allowing a greater understanding as to the effect of various foliages on the propagation of acoustic signals.

3.3 RAY PATH THEORY

A. Refraction

In a homogeneous medium, where the velocity of propagation is constant, sound energy travels in straight lines and many sound phenomena can be explained in terms of ray path theory in a manner analogous to geometrical optics. In a medium of variable velocity, Snell's law of sines (Figure 3-1) can be used to describe the refraction of sound rays; i.e. sound rays are deflected if the velocity of propagation of sound is not the same at all points. When a plane wave passes obliquely from a medium of higher velocity, c_1, to one of lower velocity, c_2, the ray will bend toward the normal to the boundary according to the law

$$c_1/c_2 = \sin\theta_1/\sin\theta_2$$

where the θ's are the respective angles relative to the normal.

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[27] M. P. Givens, W. L. Nyborg, and H. K. Schilling, "Theory of the Propagation of Sound in Scattering and Absorbing Media," J. Acoust. Soc. Am., Vol. 18, 1946, p. 284.

[28] C. F. Eyring, "Jungle Acoustics," J. Acoust. Soc. Am., Vol. 18, 1946, p. 257-270.

[29] F. M. Wiener, and D. N. Keast, "Experimental Study of the Propagation of Sound Over Ground," J. Acoust. Soc. Am., Vol. 31, 1959, p. 724.

[30] T. F. W. Embleton, "Sound Propagation in Homogeneous Deciduous and Evergreen Woods," J. Acoust. Soc. Am., Vol. 35, 1963, p. 1119.

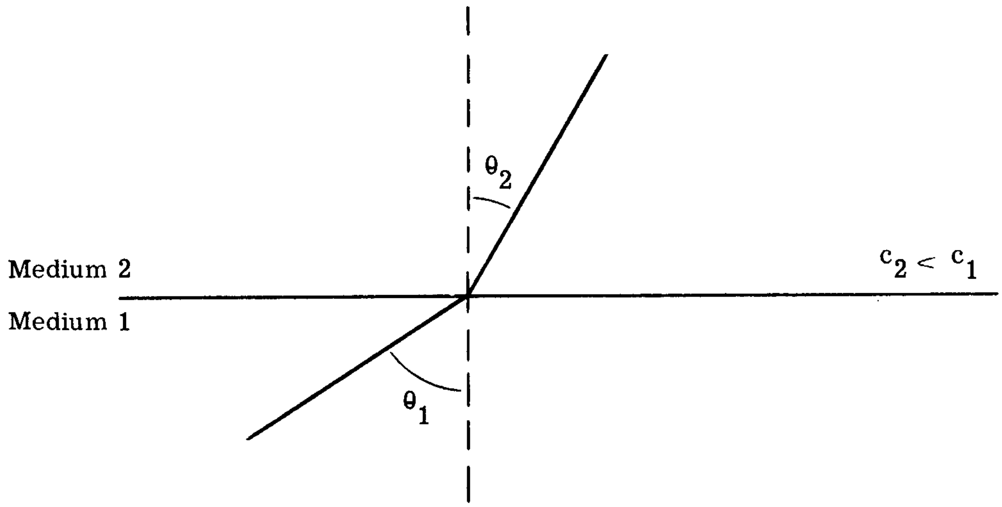

Figure 3-1. Snell's Law: $c_1/c_2 = \sin\theta_1/\sin\theta_2$

When the velocity of sound at each point in the medium is known, it is theoretically possible to calculate the sound rays, or paths, along which the sound energy propagates.

B. Effects of Temperature and Wind

In an atmosphere at complete rest, the dependence of sound velocity c on temperature is given by[31]

$$c = c_o\sqrt{T/T_o} \qquad (3\text{-}4)$$

where T is the absolute temperature in degrees Kelvin, and c_o is the sound velocity at the absolute temperature T_o.

[31] A. B. Wood, A Textbook of Sound, G. Bell & Sons, London 1946, p. 247.

If wind is present (see Figure 3-2), the resultant velocity is then the vector sum of the wind velocity $\overline{v}$ and $\overline{c}$, the latter vector being taken perpendicular to the wavefront.

C. Temperature Gradients

Assuming that the atmosphere is stratified so that the temperature at all points at the same height above ground is the same, the sound rays will be bent upward or downward dependent upon whether the vertical temperature gradient is negative (called temperature lapse) or positive (called temperature inversion). The refraction of sound through atmosphere with a constant temperature lapse rate is illustrated in Figure 3-3. The geometrical construction of the ray paths for a given velocity gradient can be found in the literature.[32,33] It can be shown that when the temperature gradient is constant, the rays are circular. All these circles do not have the same radius but all their centers are at the fictitious level of absolute zero temperature. When the temperature gradient is negative upward, this level is above the ray, which consequently curves upward as shown in Figure 3-3. When the temperature gradient is positive upward, the zero level is below the ray, and it curves downward. Temperature refraction in the atmosphere is analogous to the optical phenomenon of mirage.[34] In the case of negative temperature gradient, there is a limiting ray which just grazes the ground at a distance x from the source of sound, i.e., it is possible to have a "shadow zone" into which no direct sound can penetrate. Shadow zones are never sharp in the sense of light because acoustical diffraction effects are associated with much longer wavelengths and, in addition, sound energy is scattered into the shadow zone by turbulence.

[32] C. B. Officer, Introduction of the Theory of Sound Transmission, McGraw-Hill Book Co., Inc., New York, 1958, pp. 58-61.

[33] L. E. Kinsler and A. R. Frey, Fundamentals of Acoustics, second ed., John Wiley & Sons, New York, 1962, pp. 464-468.

[34] See p. 323 in footnote 31, p. 3-5.

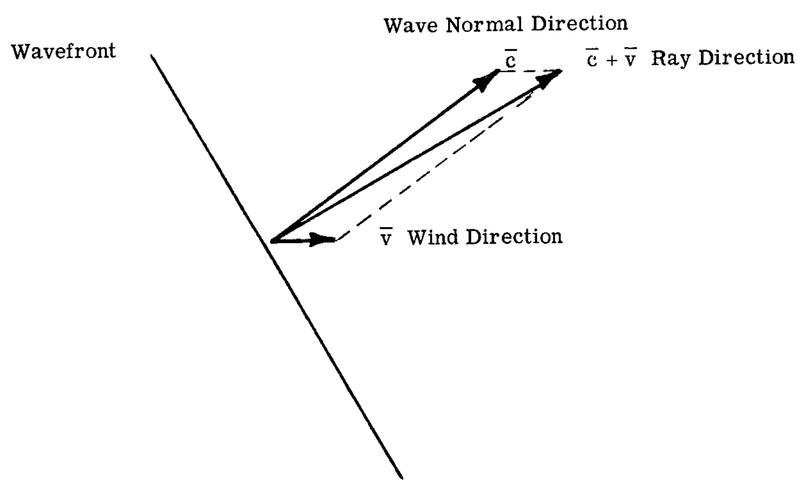

Figure 3-2. Construction of Ray Direction

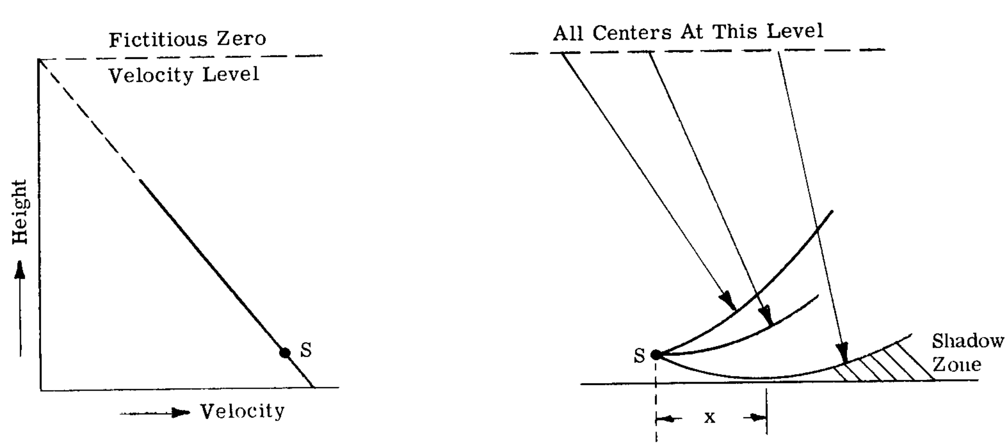

Figure 3-3. Sound Rays in a Medium of Negative Temperature Gradient

The temperature structure of open air may take various forms as shown in Figure 3-4 at three different times of day.[35]

D. Sound Channeling

When particular velocity gradients are present in the atmosphere, it is possible to create various unique situations. Consider the double gradient case in which there exists a region of increasing sound velocity with height followed or preceded by a strong decreasing sound velocity with height. If the sound source is placed in the position indicated, the ray diagram will appear as in Figure 3-5. It is observed that a level at which the velocity of propagation passes through a maximum value has a tendency to repel sound rays, and a level at which the velocity of propagation passes through a minimum value has a tendency to attract sound rays. Figure 3-6 is a temperature profile of the atmosphere that illustrates the path of sound rays leaving the source in such a way that the sound is trapped. This phenomenon is known as a sound channel and causes intensification of sound in the channel with little sound emitting outside the channel.

When sound waves are constrained in the channel, their energy is not attenuated as rapidly as with waves in a homogeneous medium. The initial spreading is spherical out to a distance r, where the channel is said to be filled, but from there on the spreading is more nearly cylindrical. Under these conditions, the transmission loss associated with spreading is given by the equation[36]

$$H = 20 \log (r_1/1) + 10 \log (r/r_1) = 10 \log r_1 + 10 \log r \qquad (3\text{-}5)$$

which is less than 20 log r, associated with spherical spreading. The low-frequency components of sound signals for explosive charges detonated in this channel can be propagated over tremendous distances.

3.4 NORMAL-MODE THEORY

The propagation of sound in an elastic medium can be

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[35] L. L. Beranek, ed., Noise and Vibration Control, McGraw-Hill Book Co., Inc., New York, 1971, p. 185.

[36] See p. 471 in footnote 33, p. 3-6.

(a) Clear Afternoon; Normal Lapse

(b) Early Morning; High Inversions With Normal Lapse Above and Below

(c) Clear Night; Ground Inversion With Normal Lapse Above

Figure 3-4. Example of Temperature Profiles at Three Times of Day

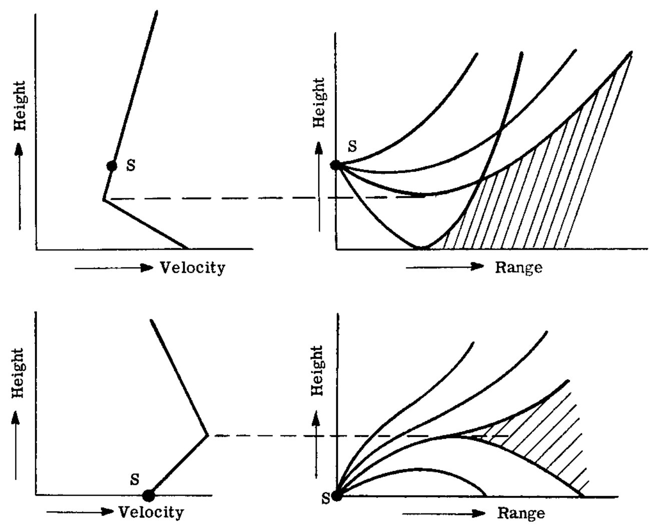

Figure 3-5. Velocity Profile and Ray Diagram

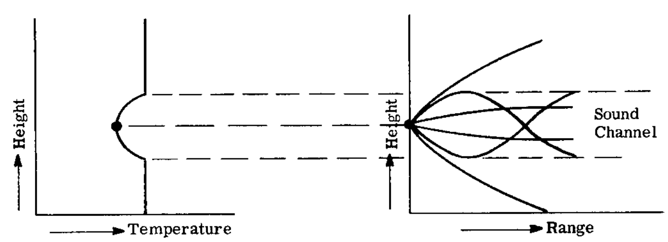

Figure 3-6. Formation of a Sound Channel

described mathematically by solutions of the wave equation using the appropriate boundary and medium conditions for a particular problem. The wave equation in rectangular coordinates may be written as

$$\frac{1}{c^2}\frac{\delta^2 u}{\delta t^2} = \frac{\delta^2 u}{\delta x^2} + \frac{\delta^2 u}{\delta y^2} + \frac{\delta^2 u}{\delta z^2} \tag{3-6}$$

where x, y, z are the Cartesian coordinates of space, t is the independent time variable, u is some dependent variable such as the local magnitude of the sound pressure, and c is a quantity which has the general significance of sound velocity and which may vary with the coordinates.

There are two theoretical approaches to a solution of the wave equation. One is the ray theory which has been summarized in Section 3.3. The other form of solution of the wave equation is the normal-mode theory, in which the propagation is described in terms of characteristic functions called normal modes, each of which is a solution of the equation. The normal modes are combined additively to satisfy the boundary and source conditions of interest.

In mathematical physics every student is familiar with the normal modes of vibration. As a rule, the concept is first introduced in the context of finite, discrete systems of masses and springs, vibrating molecules, etc. In this case it is shown that the motion of any conservative mechanical system near a configuration of stable equilibrium can be compounded of a finite number of harmonic vibrations of some specific frequencies. These ideas are extended easily enough to bounded continua, examples of which are the acoustic modes of a room, the characteristic modes of a crystal, etc. In these cases, there are infinitely many discrete eigenfunctions. When the normal mode concept is applied to unbounded continua, the examples are the treatment of the electromagnetic modes of infinite space used in the quantum theory of fields. Here the frequencies may be dense and form a continuous spectrum. Biot and Tolstoy[37] have generalized the procedure to conservative, unbounded, mechanical media of any type, so that the normal-mode

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[37]M. A. Biot and I. Tolstoy, "Formulation of Wave Propagation in Infinite Media by Normal Coordinates with an Application to Diffraction," J. Acoust. Soc. Am., Vol. 29, 1957, pp. 381-391.

concept may, in principle, be used to all types of mechanical, electromagnetic, and electromechanical waves. In certain circumstances, the normal-mode theory seems particularly suited for a description of sound propagation, for example in shallow water.[38]

In the normal-mode analysis it is noted that each different mode can exist independently of all the others, and one can thus, in principle, change the amplitude associated with a given mode without affecting any of the others. In this sense, the adjective "normal" applied to the individual modes is a true characterization of their mutual independence--quite analogous to the mutual independence of displacement along perpendicular directions (orthogonality).

To obtain a general idea about the mathematically complicated normal-mode theory, one may consider the propagation of low frequency sound in shallow water where interference effects resulting from repeated surface and bottom reflections are a dominant factor in determining the nature of the propagation. In such cases, if the bottom is reasonably flat and level, the ocean acts as a waveguide. If the wave is traveling in certain specific directions relative to the horizontal, it will be reenforced by constructive interference. In all other directions cancellation will occur. Various expressions for the normal-mode solution of the wave equation are given by different authors. Brekhovskikh[38] gives the solution for the case of a pressure-release surface and a perfectly rigid bottom as follows:

$$u = \frac{2\pi i}{h} \sum_{\ell=o}^{\infty} \cosh b_\ell z_o \cosh b_\ell z \, H_o^{(1)}(x_\ell r) \tag{3-7}$$

where

$$b_\ell = i(\ell + \tfrac{1}{2})\pi/h$$

$$x_\ell = h(b_\ell{}^2 + k^2)^{\frac{1}{2}}/h = [(kh)^2 - (\ell + \tfrac{1}{2})^2\pi^2]^{\frac{1}{2}}/h$$

$$\ell = 0, 1, 2, \ldots$$

$$k = \text{wave number} = 2\pi/\lambda$$

[38] L. M. Brekhovskikh, Waves in Layered Media, Academic Press, New York, 1960, pp. 366-385.

3.4

h = water depth

z_o = source depth

z = receiver depth

$H_o^{(1)}$ = Hankel function of the first kind

Modes propagating without attenuation are those for which the argument $(x_\ell r)$ of the Hankel function is real; those are the modes for which $kh > \pi/2$ or $h > \lambda/4$, that is, those for which the water depth is greater than one-quarter wavelength. The frequency corresponding to $h = \lambda/4$ is termed the cutoff frequency; frequencies lower than the cutoff frequency are propagated in the channel only with attenuation and are not effectively trapped in the duct. The variation of pressure with depth of the first four modes is shown in Figure 3-7.

Each mode can be conceived to correspond to a pair of waves incident upon the boundaries at an angle β and propagating in a zigzag fashion by successive reflections. For the first two modes these equivalent plane waves are shown in Figure 3-8, where the lines denote the pressure modes (zero pressure) and the + and - signs give the polarity of the pressure between the pressure modes.

For the case of sound transmission in deep sea, the normal-mode solution is different from the shallow water transmission in that deep water is considered to be underlain by an elastic medium instead of a fluid (sedimentary) medium. Adequate treatment can be found in the literature.[39] Figure 3-9 illustrates the different types of propagation paths that exist between a source and a receiver in the deep sea.[40] At short range (a), there is a nearly straight-line path between the two, and the transmission loss is determined by spherical spreading, plus a loss due to absorption if the frequency is high enough, and modified by interference effects of the surface reflection for a near-surface source and receiver.

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[39] See pp. 161-184 in footnote 32, p. 3-6.

[40] R. J. Urick, Principles of Underwater Sound for Engineers, McGraw-Hill Book Co., Inc., New York, 1967, p. 154.

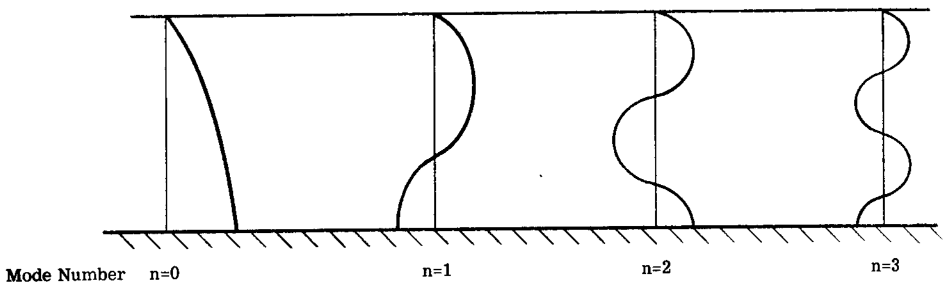

Figure 3-7. Sound Pressure Versus Depth for the First Four Modes For a Pressure-Release (Soft) Surface and a Rigid (Hard) Bottom

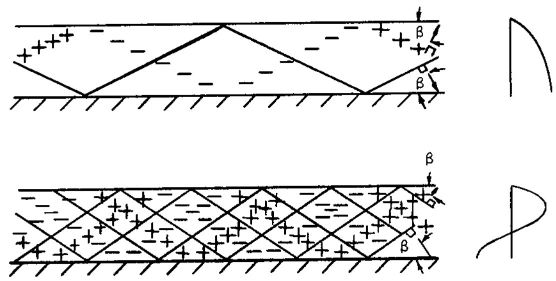

Figure 3-8. Plane-Wave Equivalents of the First and Second Modes in a Duct with a Pressure-Release Surface and a Rigid Bottom

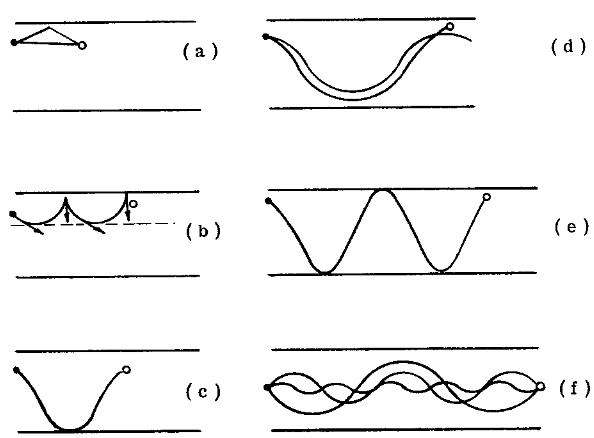

Figure 3-9. Propagation Paths Between a Source and a Receiver in Deep Water

At longer ranges, propagation occurs in the mixed-layer channel (b), involving repeated surface reflections and leakage out of the channel to a receiver below. Propagation may occur also via "bottom-bounce" reflected paths (c), which suffer a reflection loss at the sea bottom and along which the transmission loss is determined by spherical spreading plus absorption and by the bottom-reflection loss. At still greater ranges, convergence-zone transmission (d) takes place where, within a zone a few miles wide, convergence gains between 5 and 20 dB occur, depending on the depth. Beyond the first pair of convergence half-zones, the transmission between a shallow source and a shallow receiver involves multiple reflections from the bottom (e). In the deep sound channel, long-range propagation occurs over internally reflected paths (f) and is especially good when both source and receiver lie on the channel axis.

3.5 SCATTERING AND REVERBERATION

When a sound wave encounters an obstacle, some portion of the wave is deflected from its original course. It is usual to define the difference between the actual wave and the undisturbed wave--which would be present if the obstacle were not there--as the "scattered" wave. When a plane wave, for instance, strikes a body in its path, in addition to the undisturbed plan wave there is a scattered wave, spreading out from the obstacle in all directions, distorting and interfering with the plane wave. If the obstacle is very large compared with the wavelength (as it usually is for light waves and very seldom is for sound), half of this scattered wave spreads out more or less uniformly in all directions from the scatterer, and the other half is concentrated behind the obstacle in such a manner as to interfere destructively with the unchanged plane wave behind the obstacle, creating a sharp-edged shadow there. This is the case of geometrical optics; in this case the half of the scattered wave spreading out uniformly is called the "reflected" wave, and the half responsible for the shadow is called the "interfering" wave. If the obstacle is very small compared with the wavelength (as it often is for sound waves), all the scattered wave is propagated out in all directions, and there exists no sharp-edged shadow. In the intermediate cases, where the obstacle is about the same size as the wavelength, a variety of curious interference phenomena can occur.

3.5

In the study of propagation characteristics of acoustic energy, one common sound source is an explosive which contains a small charge of TNT and detonates at not very high altitude. After detonation, a shock wave that propagates in all directions is produced in the medium. A typical pressure signature near the explosion is shown in Figure 3-10 which

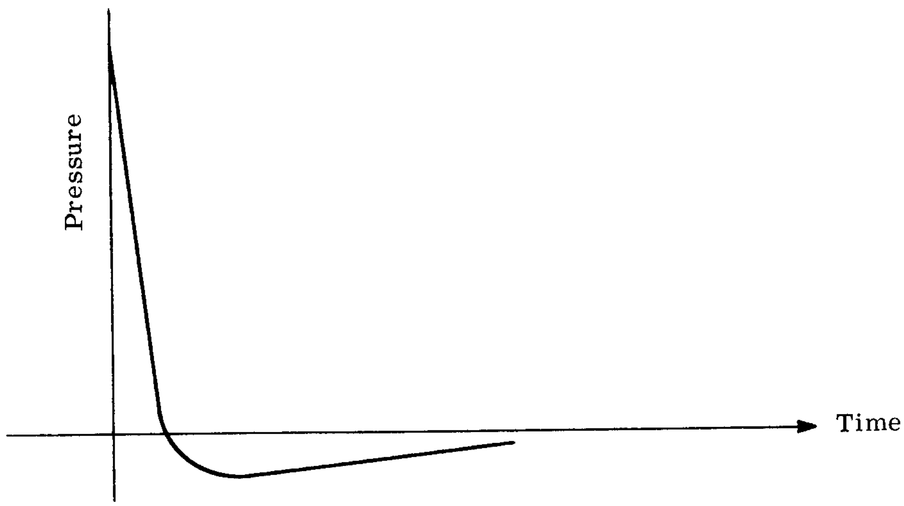

Figure 3-10. Typical Pressure Signature Near 1.5-gram Charge of TNT

has an infinitely steep front, a high peak pressure, and a rapid decay. The shock wave is normally followed by a series of minor pressure pulses due to scattering. Depending on the range from the explosive source, the received signature is complicated by refraction and multiple path in the medium of propagation. When the experiment is performed in wooded areas, the result can be expected to be very complex indeed. Some typical data is shown in Figures 3-11 and 3-12.

In a sense, the scattering of sound may be regarded as a reradiation of acoustic energy by the scatterers; the sum total of the scattering contributions constitutes the so-called "reverberation." There are two basically different types of reverberation. One type of scatterer occurs in the volume of the medium and produces volume reverberation. Trees and bushes and other foreign bodies are all responsible for the inhomogeneous structure of the air itself. The second type, surface reverberation, is produced by the stratified layers of atmosphere due to temperature or other meteorologi-

cal factors. Reverberations are heard as a long, slowly decaying, quivering tonal blast following the original blast. The theory of scattering of sound waves is extremely complicated[41,42] beyond the scope of this chapter. Because the direction of scattering is normally back toward the source, the term back-scattering is often used in practice. Whereas the attenuation and absorption of sound is of interest to most acoustics engineers, back-scattering is of importance to communication problems which involve ranging techniques. It must be taken into account under these circumstances.

3.6 SUMMARY OF FIELD RESULTS

A. Description of Test Procedures

To determine the effect of various jungle foliages on the propagation characteristics of acoustic energy in the air, a series of controlled tests was conducted in north Florida, the Panama Canal Zone, Lousiana rice fields, and the Thailand jungles during the period of time between 1966 and 1970. In general, a well controlled omnidirectional explosive sound source, which assured repetitiveness, was used with microphones placed at various distances from the source in the medium which was essentially uniform. The various ground-cover conditions are grouped as follows:

1. Heavy elephant grass, a tall undergrowth from six to eight feet in height which could limit forward visibility to one or two feet, with slender reeds, densely packed. Such foliages were found at Panama and Southeast Asia.

2. Heavy double canopy, a typical equator-like rain forest where palm trees were densely packed, with a penetrating visibility of about 50 feet, such as encountered at a typical site in Panama.

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[41] Dept. of the Navy, "Echoes, Scattering, and Reverberation," Principles and Applications of Underwater Sound, Ch. 5, 1968.

[42] L. L. Beranek, "Disturbance of Plane Sound Waves by Obstacles and by Finite Baffles," Acoustic Measurements, Ch. 3, John Wiley & Sons, New York, 1949

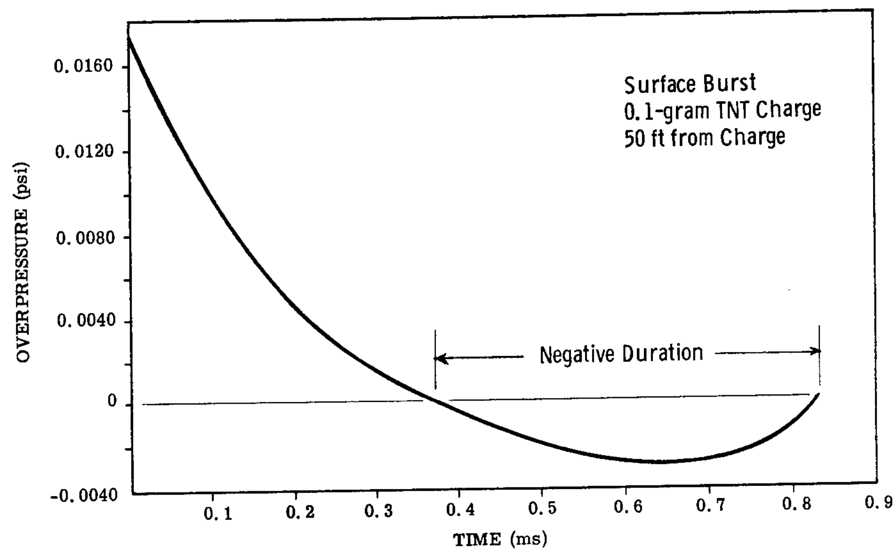

Figure 3-11. Pressure / Time for Ideal Shock Waves

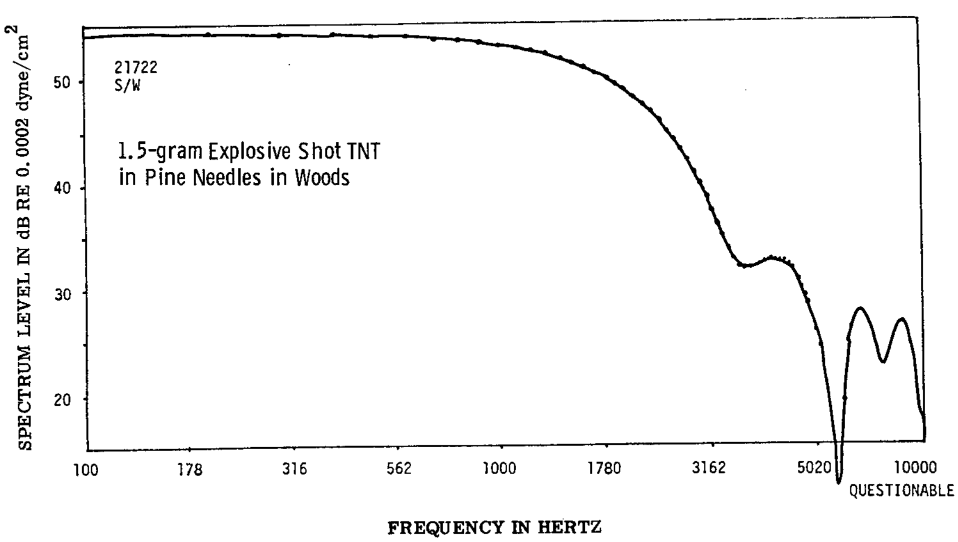

Figure 3-12. Spectrum Level of Shock-Wave Portion of 1.5-Gram Charge of TNT Recorded in Woods

3. Light forest with small leaves, typical of pine tree forests in north Florida. Here the presence of tree trunks is the predominant obstacle in the acoustical path with relatively small absorption due to the absence of large leaves.

4. Clear space, essentially light wooded area or completely open space where, in the absence of vertical temperature gradients, the propagation is influenced by the medium and the spherical spreading loss.

The tests were done with explosives containing 1.5-gram charge of TNT, placed in the foliage about 25 feet off access trails. They were detonated on a hard firing base by impact. The microphones were located at varying distances suspended off the ground, and free of any immediate obstacle.

The sound measuring equipment and test technique have been described in detail in Chapters 1 and 2. Simultaneous tape recordings were made from the various microphones which had been calibrated so that the recorded signals could be evaluated in absolute levels. As a further precaution, 10-dB step attenuations were incorporated to insure that no overload occurred in the recording system.

The collected data was analyzed both in the time domain and the frequency domain as a function of distance from the explosive charge. From the findings of hundreds of experiments performed in this project an insight as to the effect of the nature of propagation characteristic was empirically derived. These results are summarized in the next section.

B. Summary of Field Results

1. Propagation Losses

The field data was analyzed on the basis of the mathematical model given by the following equation as explained in Chapter 2, pp. 95-97:

$$SPL = A(f) - B(r,f)\log_{10} r - \alpha(f) r \qquad (3\text{-}8)$$

where A, B, and α are coefficients that are dependent upon the bracketed quantities, range r and frequency f. The attenuation coefficients $\alpha(f)$ are calculated and summarized in Table 3-1, for frequencies from 50 Hz to 2000 Hz. In general, the

Table 3-1. Attenuation Coefficients as a Function of Frequency and Foliage

CONDITIONS / FREQUENCY IN HERTZ	ATTENUATION COEFFICIENTS α, dB per foot			
	Heavy Elephant Grass 6 ft Microphone Height	Double Canopy Palm Trees 6 ft Microphone Height	Lightly Wooded Florida Pine Forest 30 ft Microphone Height	Clear Space (above river) 1 ft Microphone Height
50	0.04 - 0.05	0.02	----	----
100	0.04 - 0.05	0.02 - 0.025	----	----
200	0.04 - 0.05	0.02 - 0.03	0.01	0.005
500	0.04 - 0.05	0.02 - 0.03	0.016	0.008
1000	0.05 - 0.055	0.02 - 0.03	0.02	0.008
2000	0.06 - 0.065	0.04	0.025	0.017

NOTE: 1. When a range of experimental values was obtained the range limits are shown
2. 20 log r spreading loss assumed

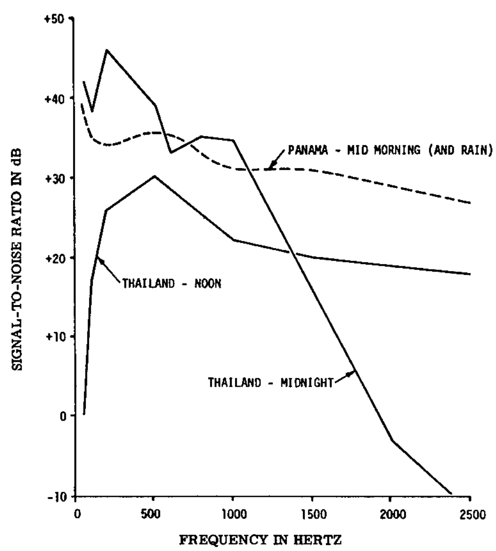

Figure 3-13. Signal-to-Noise Ratio Versus Frequency for 500 ft Distance and Various Background Noises; Source 1.5-Gram Explosion TNT

3.6.B

results are in agreement with other investigators, for example, the results of Eyring.[43] The spreading loss coefficient B in this analysis was taken as 20 for spherical spreading, which is strictly valid for a point source in a homogeneous, unbounded medium.*

It is noted from Table 3-1 that different microphone heights were used for the collection of data, depending on conditions. During the test there were indications that, in the presence of thick foliage, the attenuation is greater with lower microphone positions above ground than with the high microphone position. In clear space the spreading loss, as a function of distance and source component frequency, appears to some degree to be dependent upon the refraction caused by temperature and wind gradient. The height of the microphone off ground can make quite a difference in its reception of the sound energy.

This observation suggests that in studies of this nature, the microphone position should be related to the application for which the data will be used.

2. Ancillary Results

Figure 3-13 portrays the signal-to-noise ratios versus frequency at 500 feet for the source signal in different background noises. This normally maximum expected detection range was chosen because a design optimization in bandpass is more critical at greater ranges.

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[43] See footnote 28, p. 3-4.

*When the medium has plane-paralleled upper and lower bounds, as explained in connection with channel formation, the spreading of waves is no longer spherical because sound cannot cross the bounding planes. Beyond a certain range, the sound-pressure level radiated from the source acts as if it were distributed over the surface of a cylinder having a radius equal to the effective range and a height equal to the separation of the two parallel planes. Then the inverse first-power relation applies; i.e., a loss of 3 dB (instead of 6 dB) per doubling of the range. Unpublished results at NADC validate the magnitude of spreading loss coefficient variation in practical cases ranging from 10 to 20.

The time domain evaluation of the propagation characteristics of explosive signals was obtained through the oscillograms of amplitude versus time at various distances from the source. Figures 3-14 through 3-16 are typical of the result for small explosive shots of 1.5 grams of TNT (at 50 ft and 500 ft for two different test sites). This data provided guidance for the design of pre- and post-detection filtering and the desired AGC constants.

The effects of predetection filtering and post-detection filtering on signal plus noise are depicted in Figures 3-17 and 3-18, respectively.

The time domain pictures of some of the background noises are included in Figure 3-19. The pressure spectrum level versus frequency curves are given in Figures 3-20 and 3-21.

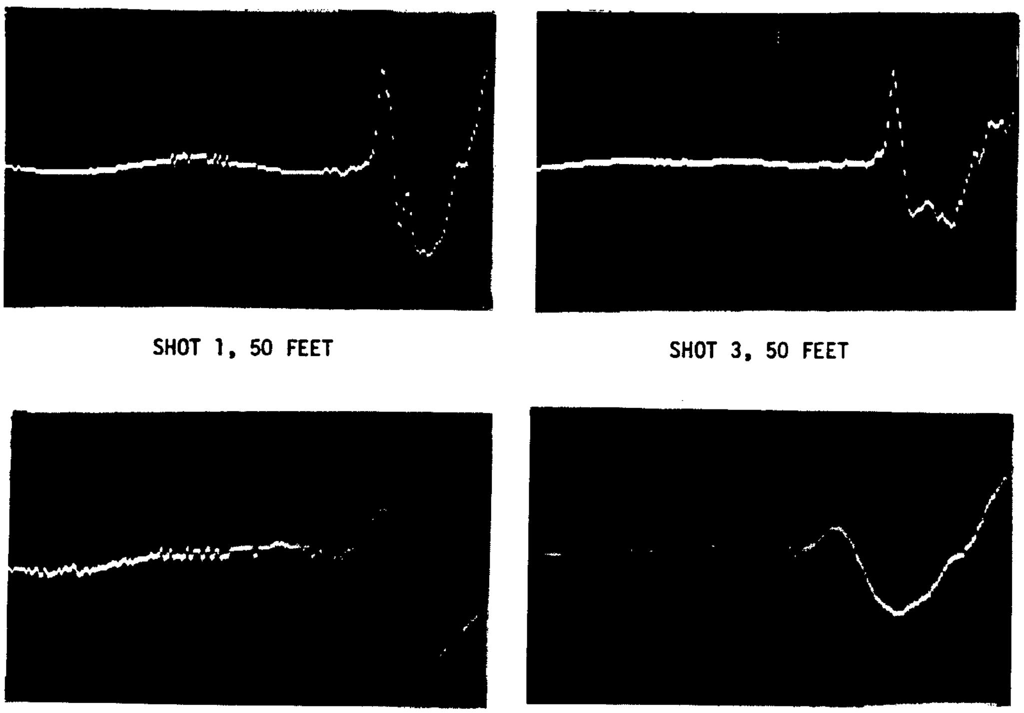
It was observed during AWG studies that the rise and decay time envelopes of the impulsive signals are significantly affected by the nature of the medium and the distance from the source, probably due to reverberation and scattering.

3.7 DISCUSSION AND CONCLUSIONS

A. Attenuation Coefficients and Time Constants

Refraction due to wind and temperature gradients can modify ordinary spreading losses, but no corrections were necessary for the data presented here. As expected, high frequencies are attenuated more than low frequencies. It must be emphasized that the separate effects of leaves, stems, and ground on the propagation of acoustic energy are not simply linear in nature, since there will always be some interaction between parts; for example, multiple reflections between the bottom of the canopy and the soil in the case of sparse undergrowth and between contiguous leaves and stems in the case of brush. These interactions are very difficult to evaluate either by experimental or analytical means. However, intuitively it seems reasonable that these interactions are small and can be ignored to first approximation. Generally, leaf area and accompanying stems will increase attenuation, especially at high frequencies. Again, the increase in the attenuation coefficient with plant density is not linear.

Figure 3-13 is useful in determining the acoustic bandpass in order to optimize the S/N ratio of a system. It is



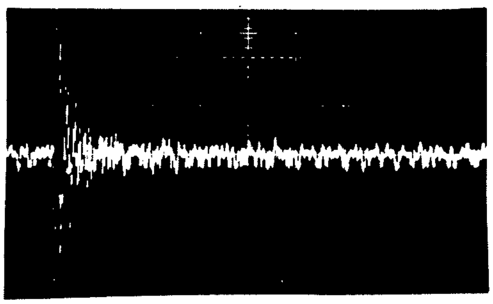
SHOT 1, 50 FEET

SHOT 3, 50 FEET

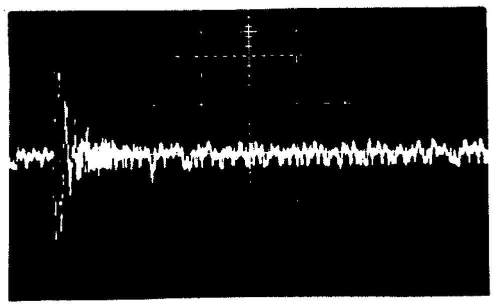
SHOT 1, 500 FEET

SHOT 2, 500 FEET

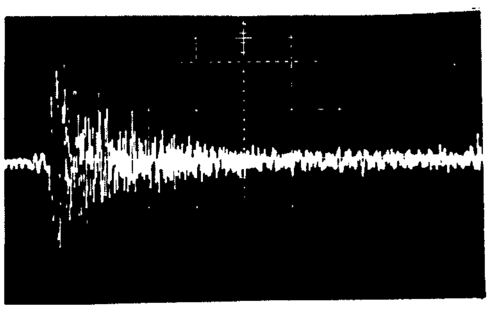
Figure 3-14. Amplitude Versus Time Presentations (1 ms/cm); Frequency Limits 50 to 5000 Hz



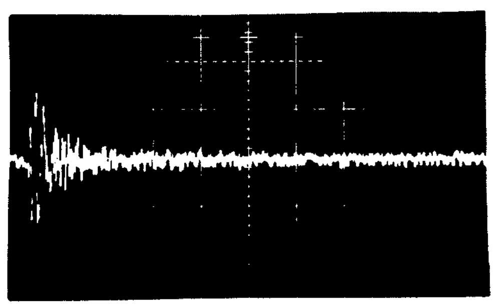
SHOT 1, 50 FEET



SHOT 3, 50 FEET

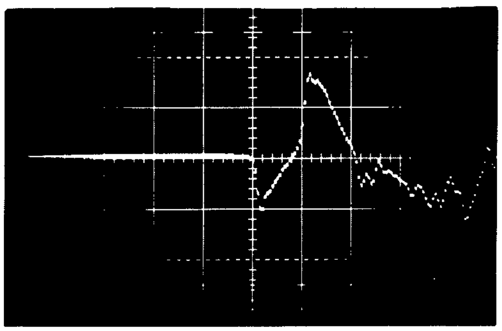


SHOT 1, 500 FEET

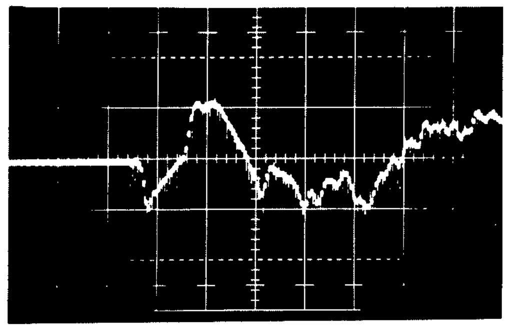


SHOT 2, 500 FEET

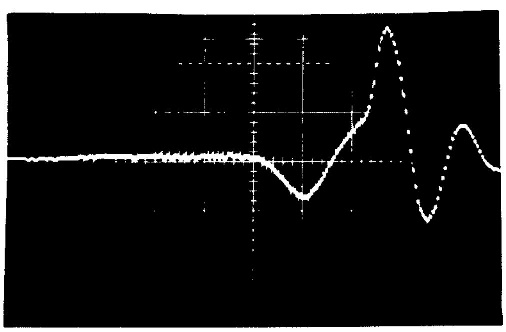
Figure 3-15. Amplitude Versus Time Presentations (50 ms/cm); Frequency Limits 50 to 1000 Hz



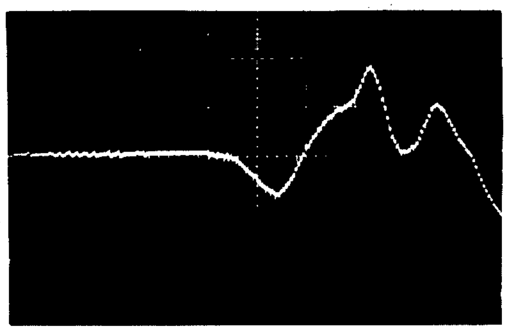
SHOT 1, 50 FEET



SHOT 2, 50 FEET

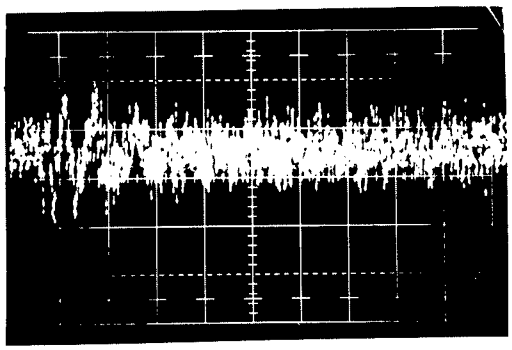


SHOT 1, 500 FEET

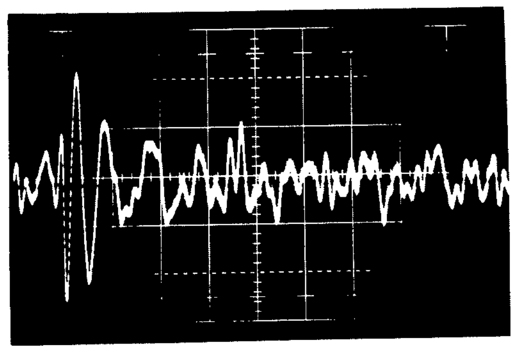


SHOT 2, 500 FEET

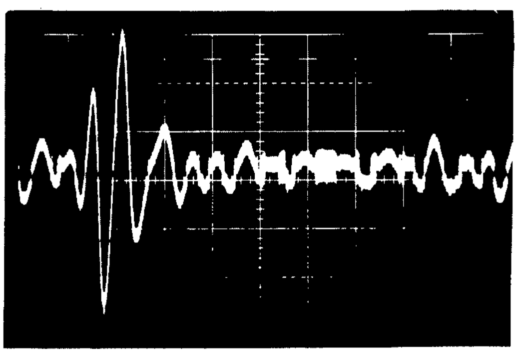
Figure 3-16. Amplitude Versus Time Presentations (1 ms/cm); Frequency Limits 50 to 5000 Hz



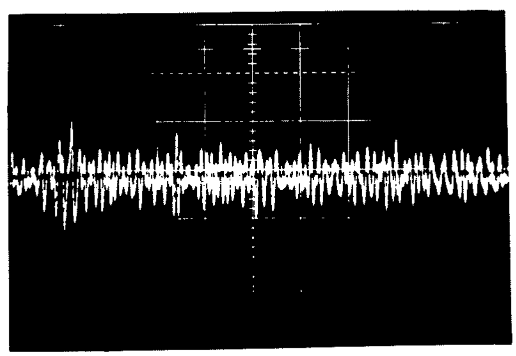
60 TO 5000 HZ



60 TO 500 HZ

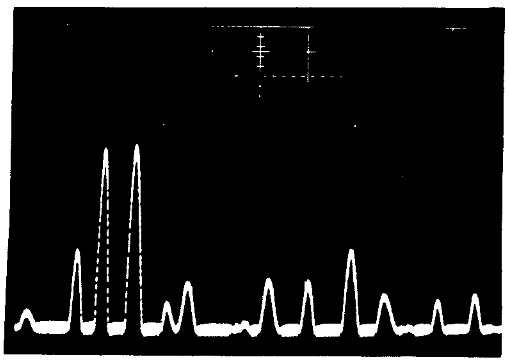


60 TO 200 HZ

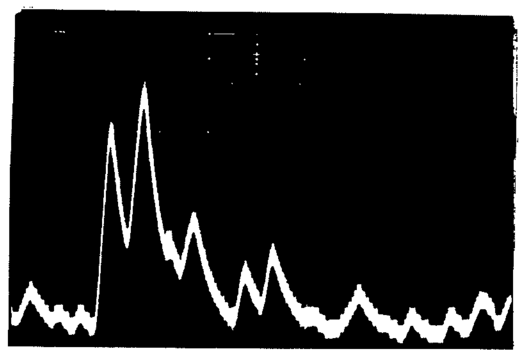


400 TO 1000 HZ

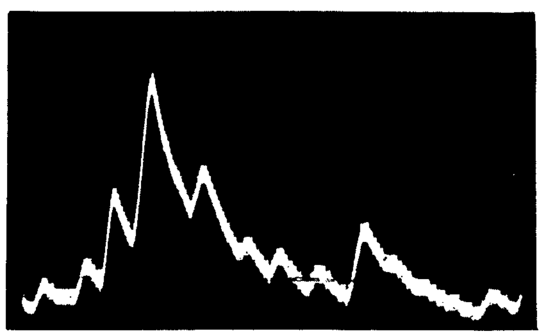
Figure 3-17. Time Domain (10 ms/cm) Presentations for Signal (No. 1 Shot, 50 Feet, West Leg, Site Two) Plus Random Noise for Different Predetection Bandwidth



NO FILTERING

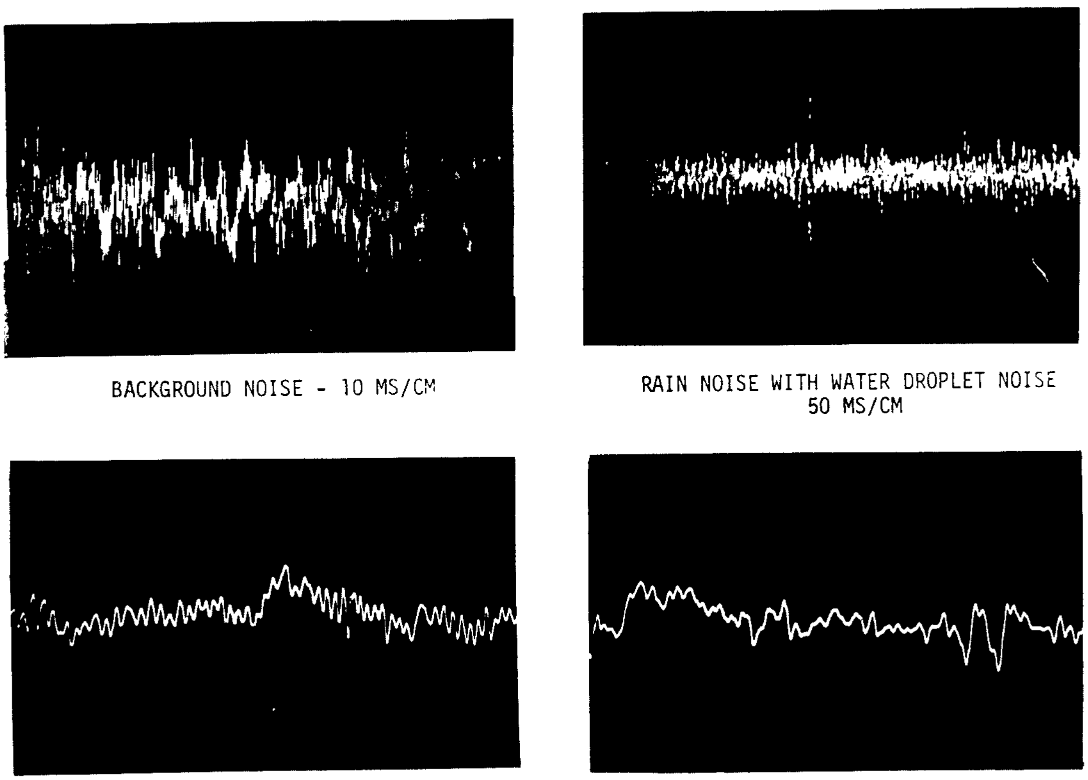


LOW PASS FILTERING OF TIME CONSTANT (RC) 2 x 10^{-3} SEC



LOW PASS FILTERING OF TIME CONSTANT (RC) 1 x 10^{-1} SEC

Figure 3-18. Post Detection Outputs for Various RC Low Pass Filters, Predetection Bandwidth 60 to 200 Hz



BACKGROUND NOISE - 10 MS/CM

RAIN NOISE WITH WATER DROPLET NOISE
50 MS/CM

BACKGROUND NOISE - 1 MS/CM

RAIN NOISE WITHOUT WATER DROPLET
NOISE - 1 MS/CM

Figure 3-19. Typical Background Noise Time Domain Presentations

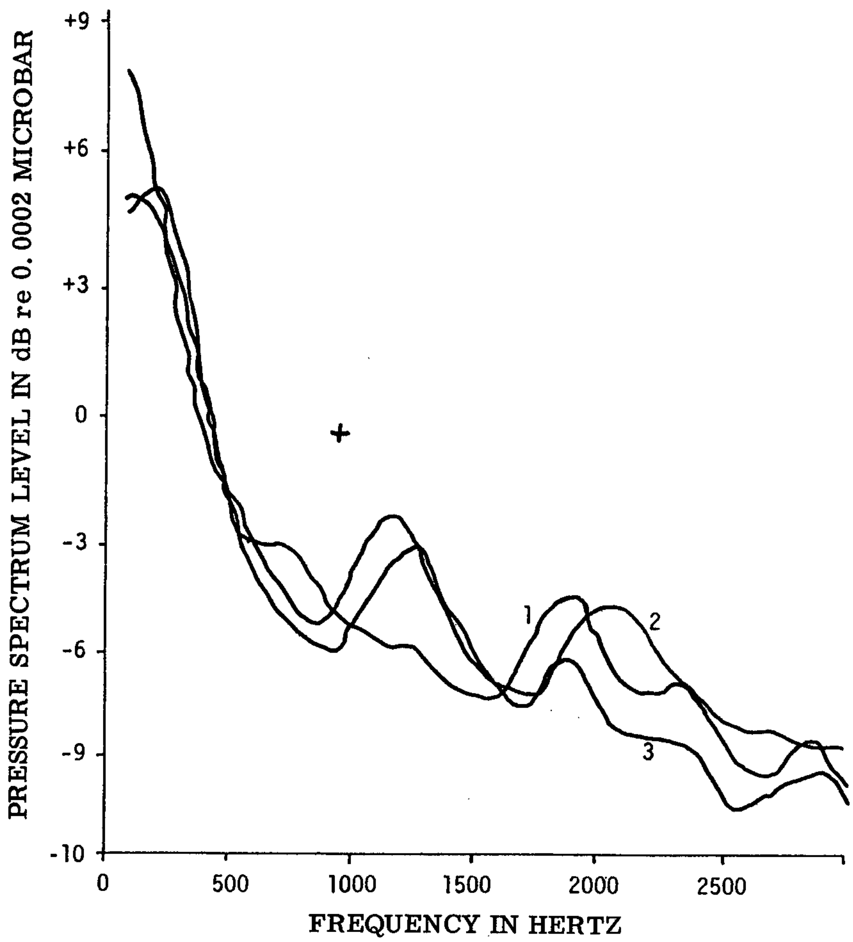


Figure 3-20. Pressure Spectrum Level Versus Frequency for Background Noise, Site 2, 100 Hz Analysis Bandwidth for Three Different Samples Within One Minute

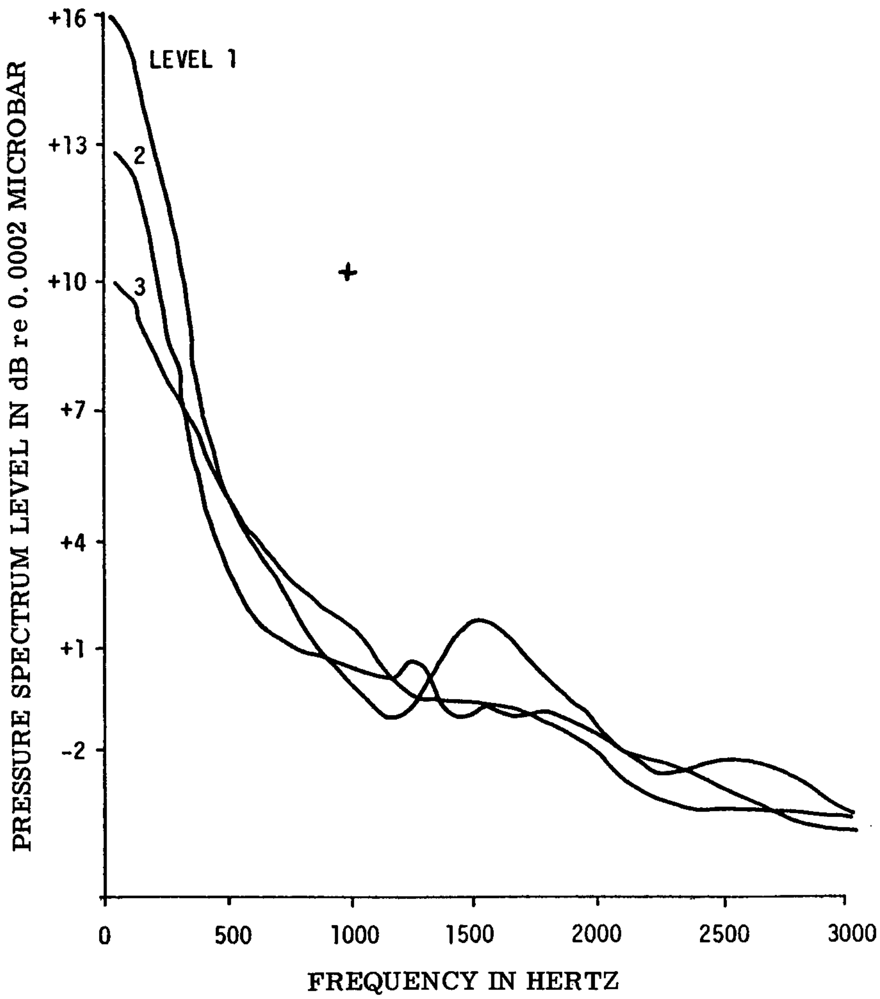


Figure 3-21. Pressure Spectrum Level Versus Frequency for Light Rain, Runs 2 and 3 Have Raindrop Noise; 100 Hz Analysis Bandwidth, Site 3

evident from the three curves that the bandpass which should be used for a near optimum S/N ratio varies considerably with the type of background noise and also probably with different media. The tests performed at various sites at different times seem to be consistent. The S/N ratio at the lower end of the frequency spectrum, say, from 50 to 1000 Hz, shows that the choice of bandwidth is not too critical.

The time domain study typified by Figures 3-14 to 3-16 reveals that the rise time of initial pulse changes appreciably with distance from the charge. From the approximate relationship that rise time is equal to the inverse of three times the upper half-power frequency, the total analysis system has about a 70 μsec capability, with the 5000 Hz bandwidth of the analyzer being the major restricting unit. The 50-foot shot oscillograms show approximately this rise time, whereas the 500-foot oscillograms have a rise time twice as large.

A further investigation of the time domain pictures shows that shot-to-shot correlation is not so predictable and greatly dependent upon range. In general, one may say that at greater ranges the envelopes have an almost exponential decay characteristic, while time constant increases considerably with range--tens of milliseconds at close range to hundreds of milliseconds at greater ranges.

B. Predetection and Postdetection Filtering

The advantage of predetection filtering was made evident by Figure 3-15 wherein the output from a random noise generator was mixed with the signal for different bandwidths 60-5000 Hz, 60-500 Hz, 60-200 Hz, and 400-1000 Hz. The optimum bandwidth reflects the point where the maximum energy of the shot occurred. The effect of post-detection filtering shown in Figure 3-18 is to improve both the threshold and signal duration characteristics as discussed in Chapter 2, subsection 2.2.H.

C. Propagation of Explosive Sounds

Classically, an explosion in free space in the time domain has a very short rise time with pressures in the nonlinear region, followed by an approximately exponential decay below the ambient pressure and a final return to ambient pressure. The frequency spectrum of this signal is continuous with increasing intensity as it approaches the lower frequen-

cies, where a broad peak is reached. As the shock wave propagates, it decreases in intensity because of spreading loss and other excess attenuation that occurs in air for finite amplitude waves. From the many tests performed on the small explosive shots of 1.5 grams of TNT in covered environment, some of the pertinent conclusions are:

1. In thick foliage, a broad energy peak between 600 and 1000 Hz at 50 feet would shift to 200 and 350 Hz at 150 feet, and to 60 and 100 Hz at 500 feet.

2. If a microphone is within 8 to 10 diameters behind or alongside an object relative to the source, the measured pressure-time characteristic exhibits a slow and irregular rise time.

3. Terrain absorption losses appear to affect the shock wave as they do the steady-state sounds.

4. Great variability has been observed in the output levels and spectra as a function of the nature of the terrain upon which the explosive charge is resting.

5. For the small explosive shot of 1.5 grams of TNT fired in the open, the spectrum decays at about 7 dB per octave above 2 kHz. For the charge fired in the woods, the decay above 2 kHz is about twice that in the open, i.e., approximately 14 dB per octave.

6. The results obtained for propagation of sound over open level terrain will not in general apply where the transmission path is predominantly in woods. This is true because in dense woods, the wind velocity and vertical wind gradients are very much smaller for a given wind velocity above the treetops than the wind velocity and vertical wind gradients over open level ground.

7. Three important parameters affecting the transmission of sound energy in wooded areas are the foliage area, trunk and limb density, and ground impedance. According to a recent paper[44], the canopy can effectively be represented

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[44]D. Aylor, "Noise Reduction by Vegetation and Ground," J. Acoust. Soc. Am., Vol. 51, 1972, p. 197

3.7.C

by a single thin wall of unknown surface area densities to sound transmission, because of multiple scattering between leaves. The formula for the loss through a thin solid wall is given by[45]

$$A = 20 \log_{10}(\pi sf/41.5) \qquad (3\text{-}9)$$

for air at 20°C, where s is the area density or the density times the thickness of the wall, and f is the frequency of the sound source. Qualitatively, this means that reducing leaf area decreases the effective wall thickness and increasing frequency increases the effective wall thickness.

As far as the size of the trees is concerned, little energy is scattered by a rigid cylinder when the sound wavelength is large compared to the cylinder radius. Thus, attenuation through forests by scattering of low-frequency sound is negligible. On the other hand, when the sound wavelength is small compared to the cylinder radius, scattering of high-frequency sound is significant.

For the effect of soil on sound propagation, softer, more porous surfaces attenuate more at lower frequency for near grazing incidence and have more specific frequency selection than harder, less porous surfaces.

[45] See p. 139 in footnote 33, p. 3-6.

CHAPTER FOUR

CHARACTERISTICS OF SOUND SOURCES

CHAPTER FOUR: CHARACTERISTICS OF SOUND SOURCES

4.1 INTRODUCTION

A. General Remarks

The preceding three chapters have dealt with the Methodology of Data Acquisition, Data Analysis and Reduction, and with the theoretical and empirical aspects of Sound Propagation. In this chapter we summarize and extract significant examples of the data acquired, analyzed, and reduced during the time frame of the AWG effort.

The results of the AWG efforts already have been reported in numerous company and agency reports. They fill many file cabinets. They were written for a number of different projects, and their objectives are not alike. Since the methodology employed evolved with time, their formats tend to differ. Many of these reports, however, contain valuable acoustical data which has broad applicability. As in the case of other "technology utilization" programs, as much of this information as possible has been brought together in unclassified summary form so that maximum use can be made of it where appropriate.

The particular sounds studied by the AWG admittedly represent a small part of the list we encounter in daily life. Some are very special and in general unfamiliar to the layman. But others have industrial and commercial importance as well as environmental significance.

All of the available AWG reports were reviewed with the intent of extracting specimen acoustical data on both natural and man-made sounds. Where formats differ, an attempt has been made to explain the differences.

4.1.A

Many of the described sounds come from different devices but have the same fundamental origin, for example, vibrating surfaces, gears, bearings, motors, fans, impact, detonations or aerodynamic and hydrodynamic effects. A brief list composed to illustrate this point is given in Table 4-1 which enumerates a few familiar categories of noisy products, and with each the corresponding basic sound-producing mechanism.

Table 4-1. Primary Sources of Noise in Well-Known Products

NOISY PRODUCTS AND DEVICES	PRIMARY NOISE SOURCES
Gas Turbines	Inlet: Fans (aerodynamic effects) Exhaust: Combustion (detonations) Vibrating surfaces
Plumbing (Valves and piping)	Impacts of vibrating mechanical parts Hydrodynamic and aerodynamic effects
Electric Motor Drives	Aerodynamic effects Bearings Gears
Air Conditioners	Fans (aerodynamic effects) Vibrating surfaces
Trash Compactors	Gears Impacts
Vacuum Cleaners	Aerodynamic effects Bearings Vibrating surfaces
Textile Machinery	Bearings Impacts Gears
Foor Processing and Packaging Equipment	Impacts Aerodynamic effects Gears
Firearms	Explosive impulse

As already mentioned, there are differences in the formats of the data, and the problem of interpretation is lessened by taking advantage of the explanations and definitions which Chapter 2 provides. The data cataloged herein is in various states of refinement or processing. These vary from completely raw to those specially treated in accordance with their mission objectives.

A compiler of acoustic data must decide whether the material is for a general readership or for program-oriented specialists. Because of this uncertainty, a general selection of data, applicable to the theme of "Technology Utilization" in civilian areas, has been attempted.

B. Presentation of Data

The reported data, having been reduced and analyzed at different points in time in a variety of ways, does not fit neatly into the Chapter 2 definitions. It becomes necessary, therefore, to interpret the data contained in the reports to relate the information to comparable forms in other sources. The improvement of this situation is one of the main objectives of the present chapter, i.e., further reduction of data to common formats.

For the purpose of identification and categorization by attributes, data may be classified in a number of ways as exemplified by the following list:

1. Source (catalog)

2. Type of Phenomenon, i.e., Signal Characteristics in Frequency Domain:
    - Ideal Signals (See pp. 2-14 to 2-18)
        - Stationary periodic
        - Stationary complex
        - Quasi-sinusoidal (phase-coherent function)
        - Stationary random
        - Transient
        - Non-stationary random

4.1.B

Real signals
- Transient
- Non-stationary random

Time Domain Characteristics (See p. 2-32)
- Envelope
- Rectified envelope
- Rise
- Fall time
- Signal duration
- Signal gradient
- Effective bandwidth

3. Place

4. Program in which derived

5. Collecting Group

6. Format

The primary data developed by the many groups involved in the AWG programs have not been reduced and analyzed to yield all of the attributes mentioned in the above list. Many of the characteristics referred to in the list, however, are to be found in the AWG reports. The various analyses most commonly encountered are described in Chapter 2 which contains examples of various forms such as One-Third Octave Analysis (p. 2-43), Spectrum Analysis (p.2-46), $\Delta f/f$ Analysis (p. 2-58), Sonograph With and Without Contouring (pp. 2-67, 2-93), 3-D Spectral Display (p. 2-69), Oscillograms (p. 2-73) and Correlation Plots (p. 2-87).

Among the most important descriptions of sounds is the spectral content which may be given in several ways. In some cases it has the appearance of line spectra. This is shown very well by the Kay Analyzer which displays spectral lines horizontally (Figure 4-1). Other displays are in the form of a frequency plot derived from a period of integration.

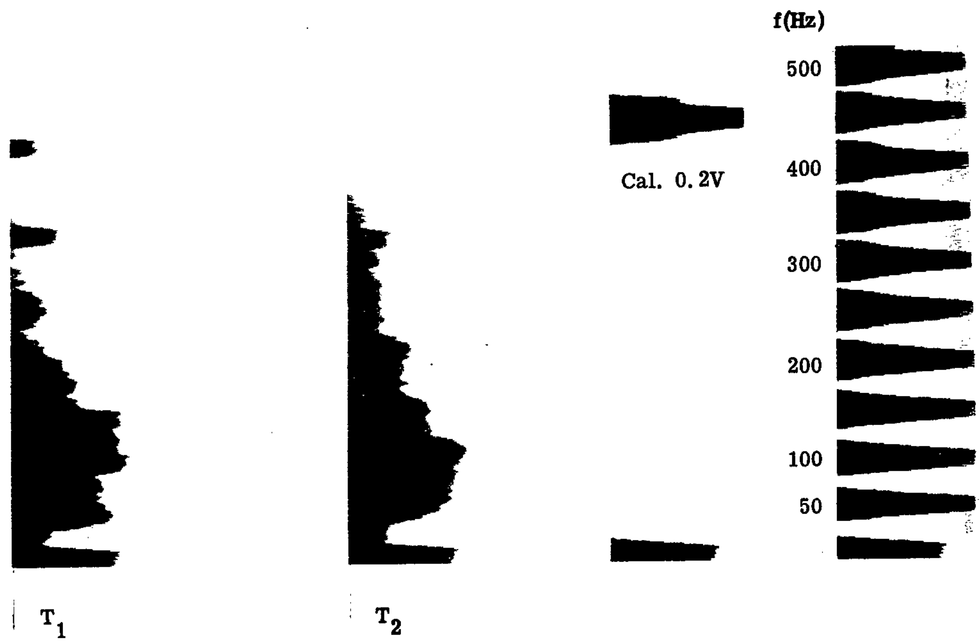


Figure 4-1. Kay Analyzer Section Spectrograms Taken at Times T_1 and T_2

By way of example, the data formats for reduced or analyzed acoustic recordings presented in NADC/AWG reports might include any of the following:

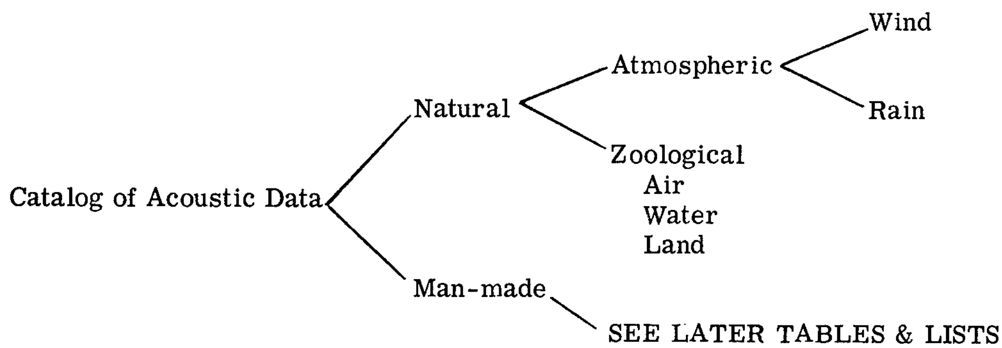
SPL plots produced by Genl. Radio 1521-B Graphic Level Recorder
Frequency Analysis plotted by 1521-B
Sonograms (Time Spectrograms)

C. Cataloging Sounds

A broad distinction has been made between natural and man-made sounds. Under natural sounds one thinks of ambients involving wind, rain, thunder, animals, insects, birds, etc. Relatively little control can be exercised with respect to these sources. With most man-made sounds (trucks, aircraft, personnel) usually there is control over orientation, range and specific operating modes, and it is possible to state absolute levels under known conditions of propagation.

4.1.C

Keeping in mind the above broad classification, the following characteristics of various sounds may be cataloged in terms of Attributes, Applications, and Instrumentation characteristics.



<u>Attributes</u>

- Spectral Data (frequency content), Line Spectra "Characteristics"
  - (Frequency dither modulation)
  - (Including explanation of background contributions)
- Time History
  - Short (Envelope, amplitude modulation)
    - (Frequency modulation)
  - Long (Occurrence)
  - Continuous vs. fluctuating sounds
    - Statistical analysis for fluctuating sound
  - Intensity
    - Source
    - Perceived (propagation as influenced by normal environment including masking factors)
    - Sound-level gradient[46] (time or distance)
    - Observation distance
    - Peak SPL
    - dB excursion
    - Initial peak SPL at 1 ft

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[46] University of Michigan, Willow Run Laboratory, "Analysis of Acoustical Surveillance Parameters," by R. F. Hand, No. 1099-5-F, Final Report, March 1969.

Forms (Signatures)
- Depends upon equipment
- Processing techniques

Properties of the sound to be cataloged
- Spectral Data
- Time history
  - Short (Envelope, modulation)
  - Long (Occurrence)
- Intensity

Applications

- Detection, location and classification
  - Techniques of display
  - Techniques of triangulation (positioning)
    - Computational
    - Analog
    - Correlation
- Masking of sounds in surveillance systems

Instrumentation

- Continuous and fluctuating sounds' intensity
  - Statistical level sorting
- Rejection of sound from particular directions.

D. Source Characteristics

Ideally the full analysis plots of frequency content or of SPL vs time would be presented. However this constitutes more detail than is desired in the type of documentation assembled here although it would provide the maximum information on the particular sound.

For the purposes of this report it has been decided that a few characteristics such as those listed below would suffice. They permit numerical tabulation rather than a requirement for graphic plots and spectrum analyzer signature formats which would constitute an Identification Manual.

In all that is reported the investigator must decide how much will be said about the surrounding acoustical environment. Either he or the reader must determine the importance of this factor and its probable influence on the data.

4.1.D

Typical Signature Characteristics

1. Main energy locations in the acoustic spectrum (lines) (as defined in pp. 2-30 to 2-33)
2. Rise and fall times
3. Signal duration (for transient types)
4. Effective bandwidth
5. Type of signal (as defined in pp. 2-14 to 2-19)
6. Frequency domain slope
7. Test Notes: air path, microphone distance, data reference, i.e., tape no., track no., location on tape
8. Frequency instability (short term) of lines

There is usually a discrepancy between the reported sound characteristic and those that would have been perceived under "ideal" measuring conditions. This difference can be charged to the influence of surroundings. One learns early in the practice of acoustics that the sound emitter, the surroundings and the listener must be considered as a system.

Our presentation, unlike the NASA survey,[47] does not attempt to explain the mechanisms of sound generation or suppression. As far as possible we merely represent the characteristics of the sound field as a source without environmental modification.

Normalization to a 1-Hertz bandwidth has been performed on some of the data. An example of normalization is given on page 2-43. Figure 2-10 is repeated here as Figure 4-2 showing the detailed corrections derived from -10 log Ff where F is the fraction of bandwidth used.

Truck signatures showing predominant lines below 62 Hz were obtained upon subsequent analysis of the recorded data. The analysis was performed with several types of sound spectrometers. Typical 6 dB contoured spectrograms having approximately 50 dB dynamic range are available for many truck operating conditions and provide the basis for the characteristic data tabulated in this report. Some analyses were made over the spectrum from zero to 640 Hz with a 0.8 Hz bandwidth filter.[48]

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[47] Athey, S. W., "Acoustics Technology -- A Survey," NASA, SP-5093 G.P.O. Washington, D. C., 1970.

[48] NADC-AWG-32, 23 May, 1969.

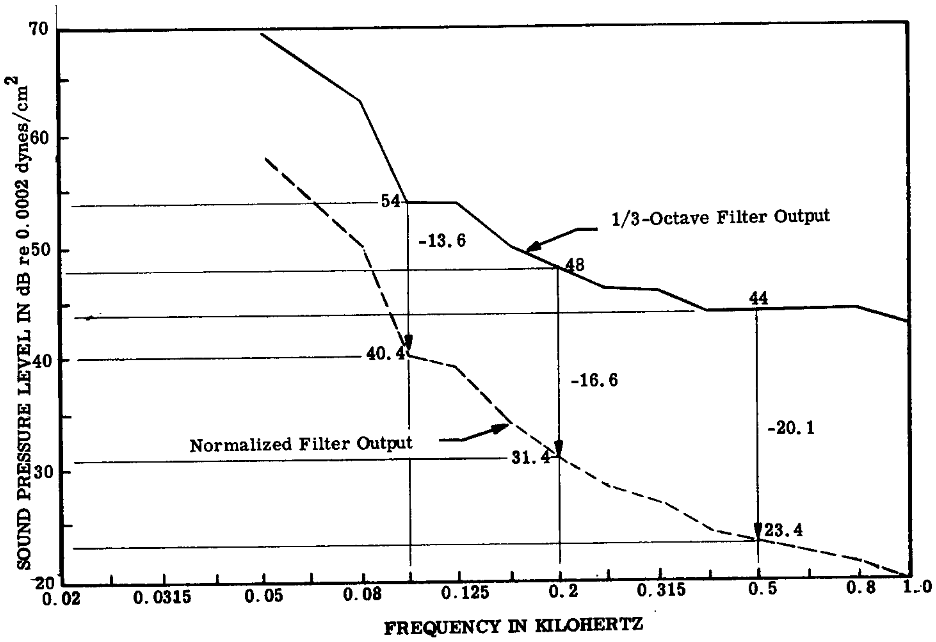

Figure 4-2. One-Third Octave Analysis of a Continuous Noise Source Showing the Effects of Bandwidth Normalization Modified by Addition of Calculated Normalization Corrections

4.2 NATURAL SOUNDS

A. General

Classification of natural sounds can be arranged according to the source, the acoustical parameters, the geographical region in which the tests were conducted, test code names (mission designation), date, etc. All of these have been used in the NADC tape library listing and cross-reference system. However, the most commonly used data retrieval entry is the "name" of the sound source. Occasionally, geographical area is of interest for studies of combat surveillance, enemy detection, etc.

Most of the acoustic data obtained by AWG was analyzed for use in surveillance. The background sounds thought to pose an interference threat to such systems were of primary interest. For example, rain has been studied because of the similarity of its spectral characteristics to pistol signatures.[49] Special filter techniques

[49] CBS Laboratories, "CBS Laboratories Dye Marker Acoustical Assistance, Analysis of CBS Tape No. 00057," Report No. 5-25-DR-012, Stamford, Conn., 14 July 1967, p. 51

have also been devised, as a result of this work, to discriminate between truck sounds and rain background.[50]

Although the specific purposes mentioned above justified the gathering of these data, the potential use of the information is much broader, extending now into civilian environmental areas.

B. Characteristics of Natural Sounds

Particular observations on certain source characteristics are given in Table 4-2 under appropriate headings.

More specific data on sounds produced by a variety of interesting animals are listed in Table 4-3.

C. Jungle Sounds[55]

Under apparently quiet conditions ambient sound pressure levels between 30 dB and 50 dB exist in the jungle, with between 2 and 5 dB per octave decrease with increasing frequency over the major part of the spectrum.

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[50] CBS Laboratories, "CBS Laboratories' Dye Marker, Acoustical Assistance Analysis of CBS Tape No. 00055," Report No. 5025-DR-013, Stamford, Conn. July 1967.

[51] Howard, J. R. and S. Krieg, "Summary Report of Acoustic Data for Phase II System," Naval Air Development Center, Johnsville, Pa., 1 October 1967, p. 2. Note: This reference contains data on filter bandwidths and AGC response time for optimum detection of the listed sounds in various ambients.

[52] NADC-AWG, Letter Report No. 22, 12 June, 1967

[53] NADC-AWG, Letter Report No. 24, 13 July, 1967

[54] NADC-AWG, Letter Report No. 28, 31 October, 1967.

[55] See pp. 1, 2, and 3 in footnote 52.

Table 4-2. Principal Sound Characteristics of Natural Sources

Rain:	50 dB peak at 60 Hz (1/10-octave analysis)[51] +4 dB/octave slope with increasing frequency 200 Hz to 2 kHz (See Figure 4-10)	
Thunder:	Signal energy primarily below 30 Hz.[51] 50 dB at 100 Hz, then -20 dB/octave 200 to 400 Hz[52]	
Wind Gusts:	Energy at 200 Hz and below 33 dB at 100 Hz, then -3 dB/octave[52]	
Ambient Jungle Sounds:	20 to 30 dB spectrum level (See Figure 4-3) -4 dB/octave slope with increasing frequency[51] (See also Figure 4-4).	
Fauna:	Insects:	0.5 to 4 kHz[51] 58 dB (20 Hz bandwidth), 0.2 kHz peaks, -20 dB/octave to 0.4 kHz, then -5 dB/octave
	Locust:	100 Hz peak, then -3 dB/octave[52]
	Bird call:	0.4 to 4 kHz and beyond[53,54]
	Frogs:	0.4 kHz
	Monkey howling:	54 dB peak at 400 Hz, secondary 35 dB peak at 850 Hz
Wind Induced:	See Figures 4-5, 4-6, 4-8, 4-9.	

Table 4-3. Specific Fauna Sound Characteristics

Type of Animal	Amount of Change in Main Excursion (dB)	Rise Time of Initial Change (msec)	Total Duration (msec)	Absolute SPL of Peak of Initial Change (dB re 0.0002 μbar)	Measurement Distance (ft.)	SPL of Peak of Initial Change at 1 ft. (dB re 0.0002 μbar)
Hornbill	17 14	80 30	600 600	54 66	100 --	97 109
Pileated Gibbon	7 10	10 20	100 100	41 54	450 --	109 121
Babblers	17 17	40 3	200 10	61 64	20 --	89 91
Barbett	--	--	2(100)	--	140	--
Owl	10	10	150	46	70	85
Bulbol (Bird)	22	20	1000	66	--	102
White Headed Gibbon	15	50	5(300)	64	270	121
Barking Deer	10	20	50	33	250	--
Toad (Kalula)	16	200	500	61	30	--
Macaca	24	40	50	94	40	122
Rana Limnocharis (Frog)	--	--	150	--	20	--

4.2.C-4.2.D

Figure 4-3 shows a quiet jungle spectrum and insect noise level (normalized to a 1-cycle bandwidth).[56] Further information on this subject is to be found in C. Eyring's publication.[57]

Figure 4-4 in contoured sonogram format, displays with even greater detail the transient spectral phenomena of a chirping jungle bird against the ambient sounds.

D. <u>Wind</u>

In our tabulation of Characteristics of Natural Sound Sources (Table 4-2), we merely list data on spectral characteristics. Wind is such an important factor in outdoor low-level measurements or in windy locations that it deserves further comment.

Chapter 1, Data Acquisition, discusses the wind effects and the use of windscreens. In addition it is noted that wind noise is a frustrating factor in critical low-frequency regions of the spectrum. Noise levels and spectral properties of wind interacting with typical commercial microphones are given in Figures 1-8 and 4-5 for various velocities. These figures further indicate the increase in wind noise with wind velocity from 10 mph to 25 mph. A difference of 15 dB is noted in this example.

Wind direction relative to the microphone has a more pronounced effect than other low-frequency sounds because of the mechanism by which the noise is generated, i.e., interaction of the air with the diaphragm enclosure and its shield which results in turbulence. Wind noises that arise from the rustling of leaves etc. should be distinguished from that above and are characteristic of the materials acted upon. A tabulation of all these would indeed be long. Usually the

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[56] University of Michigan, Willow Run Laboratory, "Optimization of Acoustic Surveillance Systems," by R. F. Hand, No. 8510-4-F, Final Report

[57] See footnote 28, p. 3-4.

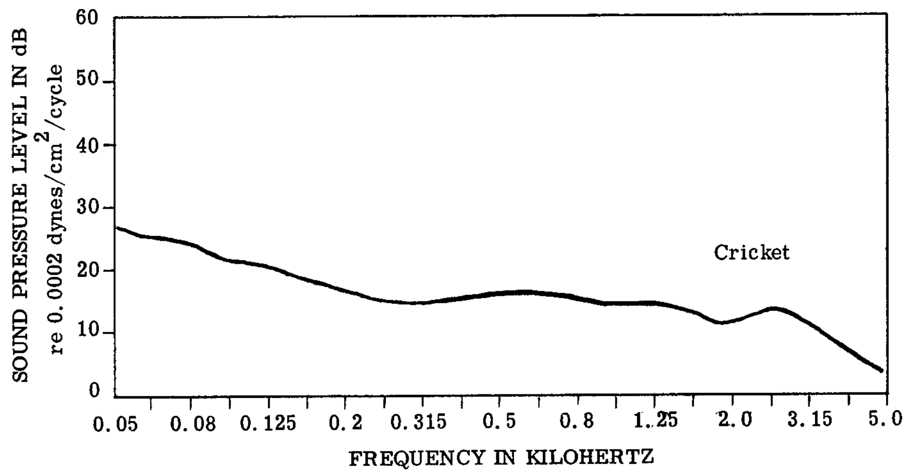


Figure 4-3. Quiet Jungle Spectrum Showing Insect Noise Level at Observation Distance (Normalized to 1 Hertz Bandwidth)*

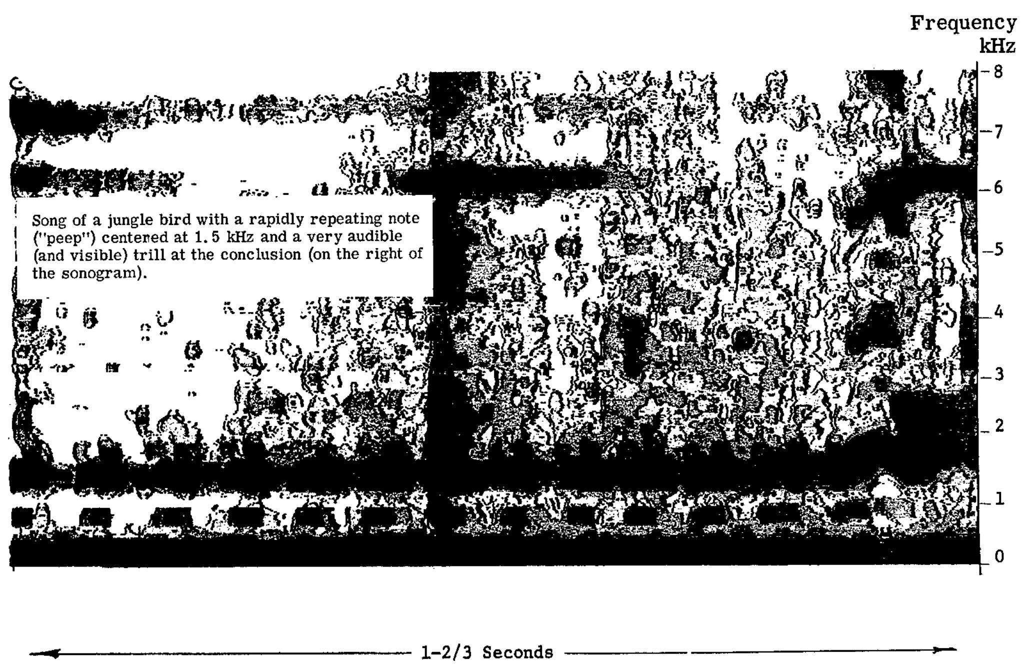


Figure 4-4. Sound Spectrometer Identification of the Frequency Location, Pulse Rate and Duration of a Distinct Chirping Sound in a Noisy Environment

*See page 4-9 concerning bandwidth normalization

commoner sources such as trees, leaves, etc. are given, but acoustical workers more frequently encounter the wind-microphone generated artifact. These observations explain why a 6 to 8 dB direction effect at low frequencies is indicated in Figure 4-6 whereas Figure 4-7 shows directionality mainly above 1 kHz.

Figures 4-8 and 4-9 indicate the fluctuation in wind noise level in a typical low-level background measurement.

E. Rain

Rain can be important in various acoustic surveillance and identification applications because it constitutes an interfering signal. Knowledge of its spectrum aids in the design of filters and other electronic measures to reduce its confusing influence.

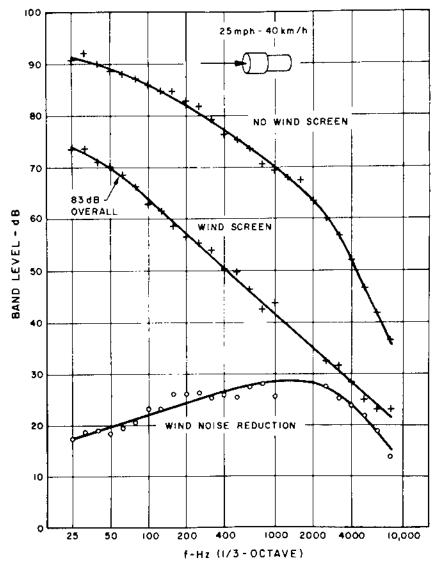
Figure 4-10 shows a 1/10-octave spectrum for rain. This curve shows both a low-frequency peak at about 60 Hz (possibly hum in the analyzer) and another at about 2 kHz. Data of this type is difficult to authenticate because the question of concurrent ambient creeps in. In other words it is impossible to do a controlled experiment in which the rain is turned on and off and thus neatly subtract out the background sounds. As a result, some of the energy displayed in Figure 4-10 is undoubtedly from sources other than rain.

4.3 MAN-MADE SOUNDS

A. General Remarks

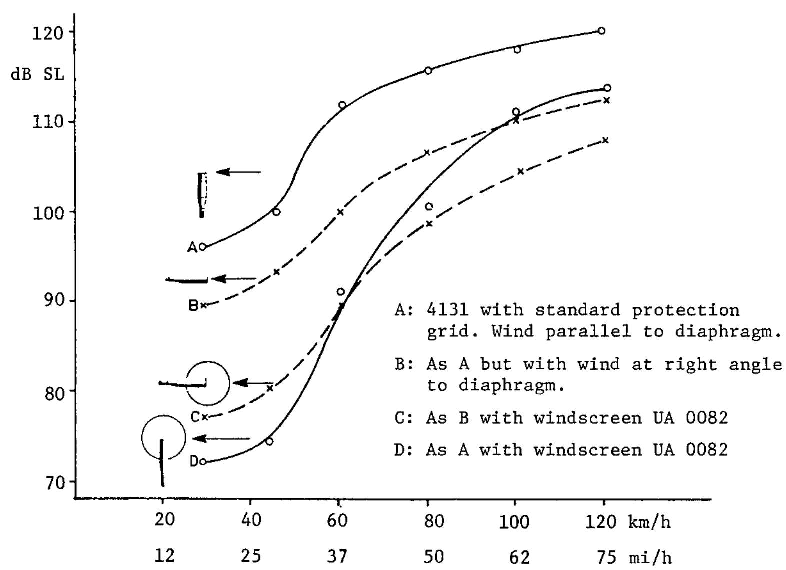
Various man-made noises were studied in certain foreign and domestic terrains. The military purposes originally intended are now applicable to a number of civilian problems. One example is vehicular noise. Acoustical techniques developed by the Navy for truck and other vehicle (land, air and water) detection, identification, classification and tracking may now be usefully applied to noise law enforcement, crime prevention, insurance loss prevention and nuisance reduction.

Field teams were sent into areas of southern and western U. S., Panama, Southeast Asia and elsewhere to military reservations where varied vehicular activity was prevalent. Recordings, carefully calibrated, were obtained together with photographic documentation and detailed terrain descriptions.



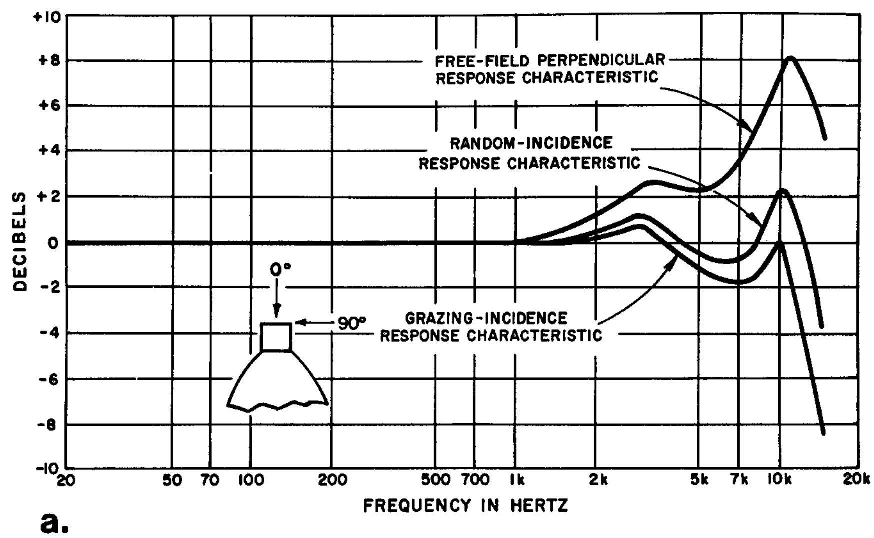
(Courtesy General Radio Handbook of Noise Measurement, 7th ed., 1972)

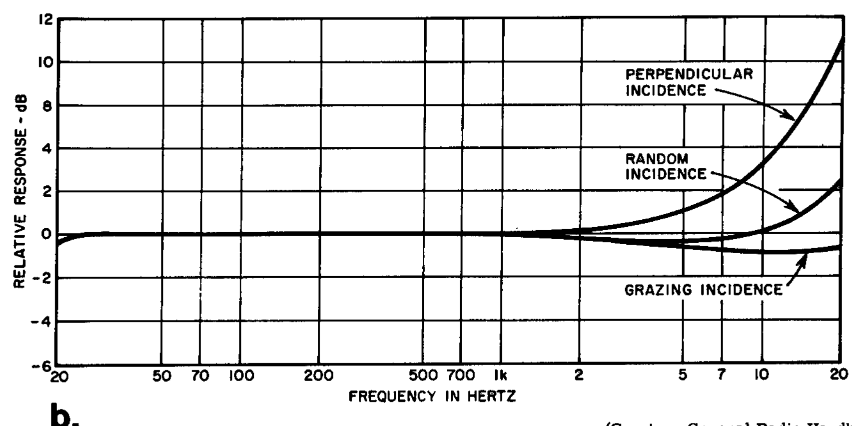
Figure 4-5. Wind Noise Spectrum



(Courtesy Brüel and Kjaer, Acoustic Noise Measurements, 1971)

Figure 4-6. Wind Noise as a Function of Wind Speed (in the Frequency Range 20 Hz-20 kHz) Measured With and Without Windscreen





(Courtesy General Radio Handbook of Noise Measurement, 7th ed., 1972)

Figure 4-7. Typical Response Characteristics for Ceramic Microphones (a) 1-in., (b) 1/2-in.

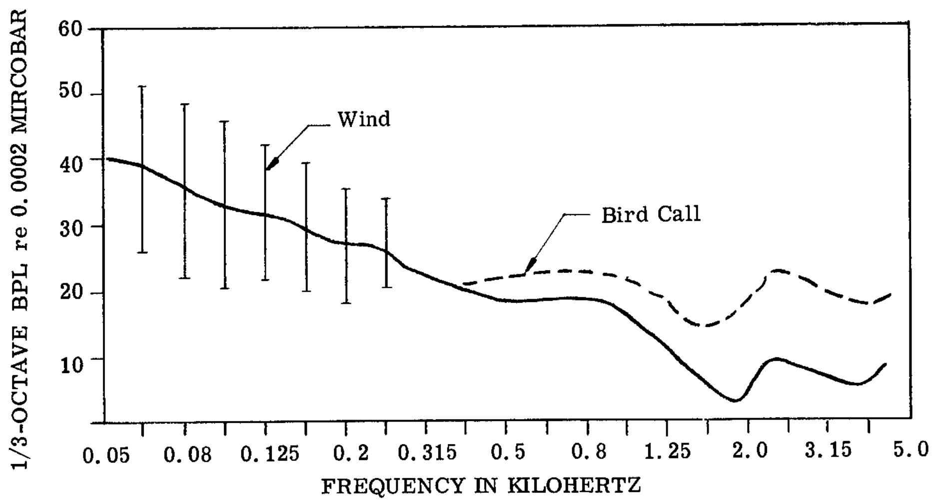


Figure 4-8. Wind Noise Fluctuations Relative to Bird Call BPL

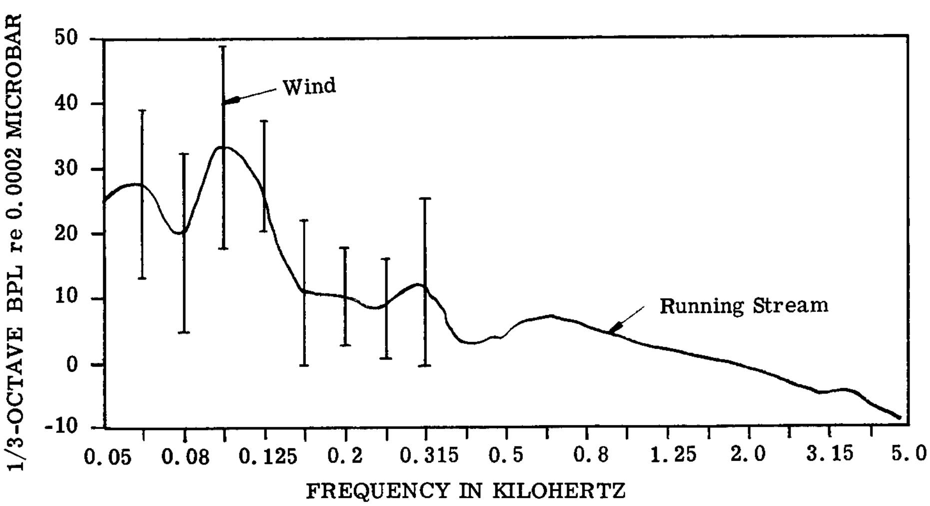


Figure 4-9. Wind Noise Fluctuations Against Background Sound of Running Stream

4.3.A

Sound analysis programs were established in military and civilian laboratories in the U. S. They produced spectral and intensity data together with special analyses involving electrically sophisticated instrumentation.

The question arises concerning the complexity of systems employing computers and whether civilian uses could justify the cost. One example of complex analysis is considered below.

A signature may consist of a relatively uniform continuous distribution of frequencies with spectral lines superimposed. Experience in detecting and classifying sounds has shown that the discrete spectral line components are most important.

In presenting data on sounds in general, whether man-made or natural, there is always the question of how detailed the description shall be. The discrete spectral line is a case in point, for we may simply note its presence or we can track it and describe its dynamics. The latter can be done using a phase-lock tracking loop. This is one way a spectral line's individual behavior can be utilized for characterization, i.e., a step beyond the mere recognition of its presence. Often this characteristic of line stability can be used to distinguish between man-made and natural sounds. The ability of the phase-locked system to track line frequency is discussed more specifically in the following subsection on truck signatures.

It should be remembered that acoustic identification of sounds may often be supplemented by seismic signals obtained at the same time. Seismic information may thus present an additional and deciding variable for discrimination purposes.

Much of the sound data collected by NADC could be termed "fly-by". That is, the characteristics of sounds generated by a vehicle nearing and passing the closest point of approach, CPA, were recorded. This information was important to the uses of the Navy's vehicle detection and identification program. Such recordings revealed the essential dynamic effects such as Doppler, multipath propagation and aspect effects. These data were useful to the Navy's need at the time but would also have to be taken into account in similar civilian uses into which the technology might be transferred. One thinks immediately of automatic or instrument-aided identification of vehicles and aircraft in an environmental noise monitoring system.

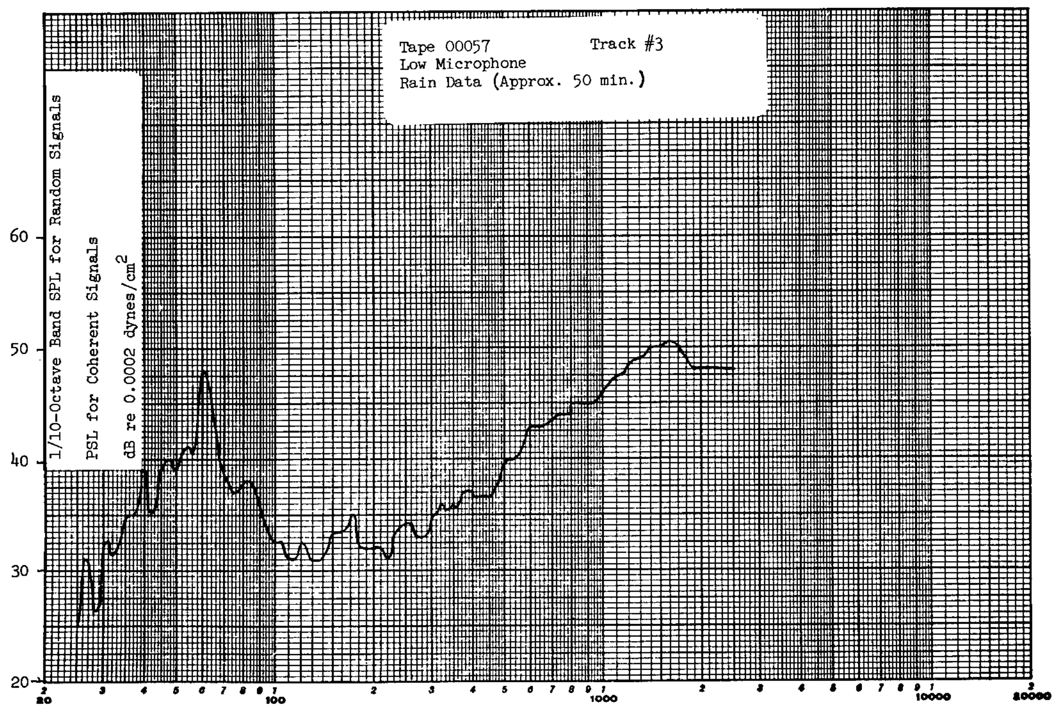


Figure 4-10. Rain Spectrum

Under technology utilization one might include the emerging interest in machine performance diagnosis. This ranges from checking troubles in the family car up through more sophisticated inspection methods used in quantity-produced consumer devices, kitchen appliances, power tools and production machinery to helicopter rotor bearings and other high-performance aircraft engine components.

In the paragraph above be have mentioned two technology transferrences, vehicular identification and machine performance diagnosis. There is a third which deserves mention in that it may be derived from some of the recorded Navy data. This has to do with propagation which was covered fundamentally in the preceding chapter. It is an item of great import to acoustical engineers who are currently assessing the expected noise levels in communities exposed to new or enlarged highways, aircraft overflight, etc. The Federal Government's Environmental Protection Agency (EPA) requires that such information be included in the so-called "Environmental Impact Statement" for each proposed undertaking. It is also part of the noise

assessment procedure recommended by the Office of Housing and Urban Development (HUD). The Department of Transportation (DOT) is similarly involved. Not only is there to be a prediction of propagated noise levels, but means of abatement and control through the selection of terrain heights, landscaping and other features are to be proposed. The Navy data may be useful here, because the influence of terrain can be derived in many cases. Some of the logs contain site descriptions which are quite accurate and hence can relate landscape influences and sound propagation. This could form the basis for a separate study using the considerable resources in the Navy library of sound recordings. In other words, it should be possible to extract the terrain influences without redoing the field tests, a great saving of effort.

B. Truck Signatures

Truck signatures are influenced by many factors including the environment, road conditions, the driver's driving habits as well as muffler design and the engine characteristics of a particular vehicle. To distinguish between different trucks by means of their acoustic signatures, use may be made of both time and frequency domain (spectral) data. However, it has been found[58] that although differences in the behavior of short-term instabilities of the time domain signature can be observed, the predominant spectral components are more useful for classification of the vehicles.

The vehicle noise sources include the body, tires, transmission, differential and engine as detailed below. These are seldom pure tones, especially in power trains which have rich harmonic content.

1. Signature Analysis

Factors included in the identification process are:

Crankshaft Rate (CSR)
Cylinder Firing Rate (CFR or CR)
Engine Firing Rate (EFR)
Predominant Harmonic Pattern
Drive Train Components (DT)
Engine Idle Signature
4th Gear Signature
3rd Gear Signature
2nd Gear Signature
Harmonic Grouping
Gear Mesh Line (M)

Vehicle engines operate between 600 and 3600 RPM with trucks near 2000. Gear shifting, unsteady throttle and rough roads which cause violent changes in load upon the engine result in spectral line jitter. This instability provides one of the major clues to vehicle identification.

Another acoustic characteristic of truck noise which can help to identify it is its slow buildup in sound pressure level which reaches a peak at closest approach and then recedes.[59] Features of a system for identifying trucks as a group, but not particular types or makes, would include sound amplification, filtering, threshold discrimination and signal duration sensing.

Filtering will eliminate energy sources outside the 100 to 500 Hz band of principal discrete frequencies produced by truck engines. The main energy is concentrated at one or two discrete frequencies around 100 Hz. It may fall as low as 80 Hz (2-1/2-ton gasoline truck) depending upon vehicle and operating conditions: example, 100 Hz for a 5-ton diesel[60] has been obtained.

A requirement for signal duration of at least several seconds at a selected threshold level will distinguish truck signatures from various other rapid transient phenomena such as noises produced by fauna and personnel. Moreover, a rate of increase and decrease of the signal, typically in excess of 0.2 dB/sec, is characteristic of truck sounds. One may also wish to place a signal-to-excess-over-background requirement of at least 6.0 dB upon the identification process.

In spite of the above distinctions we will see that aircraft sounds are sufficiently similar to be mistaken for trucks. But note the fact that human perception easily distinguishes between

[58] NADC-AWG-34, "Study of Truck Classification by Spectral Analysis," 15 July, 1969, p. 2.

[59] NADC-SD-7135A, "Land Vehicle Detection" ASW Applications of Defense Special Projects Group Sensor Technology, 9 September, 1971, p. 22.

[60] See p. 3 in footnote 51, p. 4-10.

the two. The reason for this is our remembered knowledge of spectral line dither. Both trucks and aircraft have spectrally rich signatures, but usually there is considerable instability in the former due to vehicle RPM changes. Aircraft spectral lines are comparatively more stable. Phase-locked servo loops, mentioned previously, can be devised to lock onto and track lines in such a way as to assess their jitter, thus discriminating between truck and aircraft sounds.

C. Notes on Internal Combustion Engine Signatures

From truck signatures (spectrograms) one can determine predominant frequencies which in turn identify the type of engine. In terms of Cylinder Firing Rate (CFR) and selected gear these predominances can be expressed as harmonic components. Consider four different vehicles and their significant identifying characteristics [61] (see Table 4-4 and Figures 4-11 to 4-18):

The spectrum consists of several discrete frequency components related to:

Crankshaft Rate (CSR)
Cylinder Firing Rate (CFR), one cylinder
Engine Firing Rate (EFR), all cylinders

The following basic relations may be used:

EFR = CFR x No. of cylinders
CFR = CSR (2 cycle)
CFR = 1/2 CSR (4 cycle)
Lowest frequency = CFR
Strongest line = EFR coincides with a harmonic of CFR
2nd harmonic of EFR coincides with a harmonic of CFR
Predominant Pattern = above coincidence of frequencies
Examples:
1. There is a predominant pattern of six in the six-cylinder in-line engine
2. The exhaust configuration in a V-6 produces a pattern of three

Patterns often not obvious at low order harmonics become so in the mid- to high-frequency range

[61] NADC-SD-72058, "Vehicle, Aircraft Classification Manual," 21 April, 1972, p. 5-4.

Table 4-4. Typical Listing of Predominant Identification Frequencies and Operating Gears of Four Vehicles

VEHICLE	A	B	C	D
Vehicle Weight (lbs)	1970	7917	18,230	19,230
No. of Cylinders	2 opposed	6 in line	6 in line	6 in line
Stroke	2	4	4	4
H.P.	14	94	140	200
Idle				
Predominance	None	6	6	6
Strongest line	CFR 6	CFR 6	CFR 18	CFR 6/12
Other strong line	CFR 12	CFR 18	CFR 6/12	CFR 6/12
2nd Gear				
Predominance	None	6	None	3
Strongest line	CFR 2	CFR 6	CFR 6/12	CFR 6
Other strong line	CFR 4	CFR 12	CFR 6/12	CFR 9
3rd Gear	etc.	etc.	etc.	etc.
4th Gear	"	"	"	"
5th Gear	"	"	"	"

These are typical listings useful in the identification process. Examples of the spectrograms are shown in Figures 4-11 to 4-18.

Table 4-5. Engine Spectra Formulae

	6 Cylinder		8 Cylinder		12 Cylinder	
	2 Cycle	4 Cycle	2 Cycle	4 Cycle	2 Cycle	4 Cycle
CSR (Hz) Crankshaft Rate	$\frac{RPM}{60}$	$\frac{RPM}{60}$	$\frac{RPM}{60}$	$\frac{RPM}{60}$	$\frac{RPM}{60}$	$\frac{RPM}{60}$
CFR Cylinder Firing Rate	CSR	$\frac{CSR}{2}$	CSR	$\frac{CSR}{2}$	CSR	$\frac{CSR}{2}$
EFR Engine Firing Rate	CFR x 6	CFR x 6	CFR x 8	CFR x 8	CFR x 12	CFR x 12
EFR Engine Firing Rate	$\frac{RPM \times 6}{60}$ $\frac{RPM}{10}$	$\frac{RPM \times 6}{60 \times 2}$ $\frac{RPM}{20}$	$\frac{RPM \times 8}{60}$ $\frac{RPM}{7.5}$	$\frac{RPM \times 8}{60 \times 2}$ $\frac{RPM}{15}$	$\frac{RPM \times 12}{60}$ $\frac{RPM}{5}$	$\frac{RPM \times 12}{60 \times 2}$ $\frac{RPM}{10}$

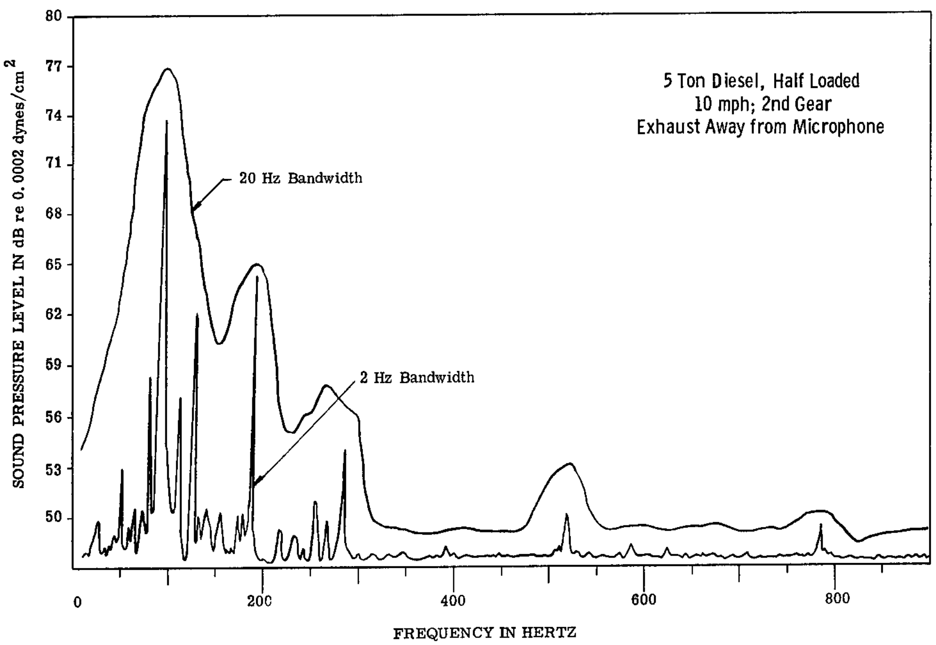


Figure 4-11. Spectrogram, Broadband and Narrow Band, for a 5-Ton Diesel Truck at 10 mph, 2nd Gear

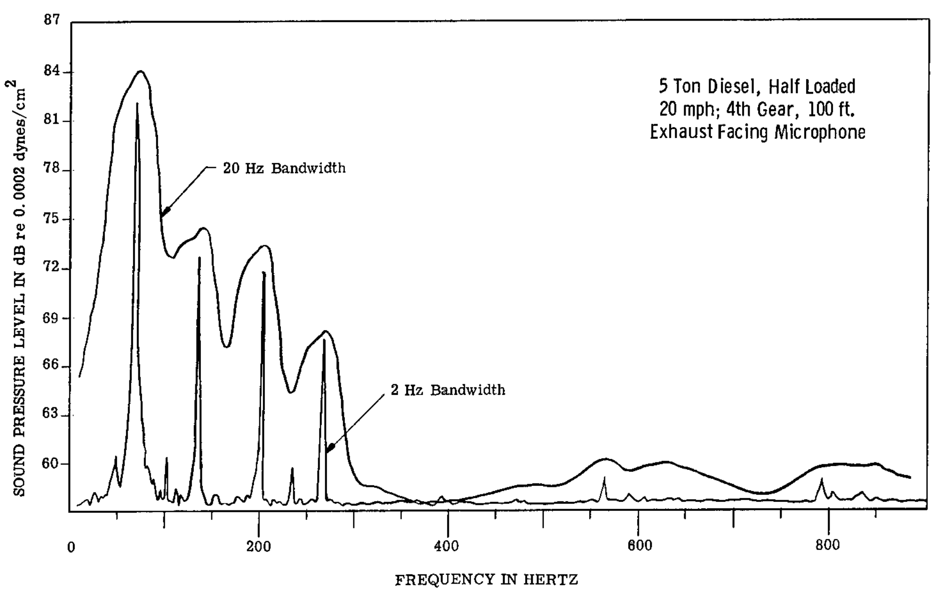
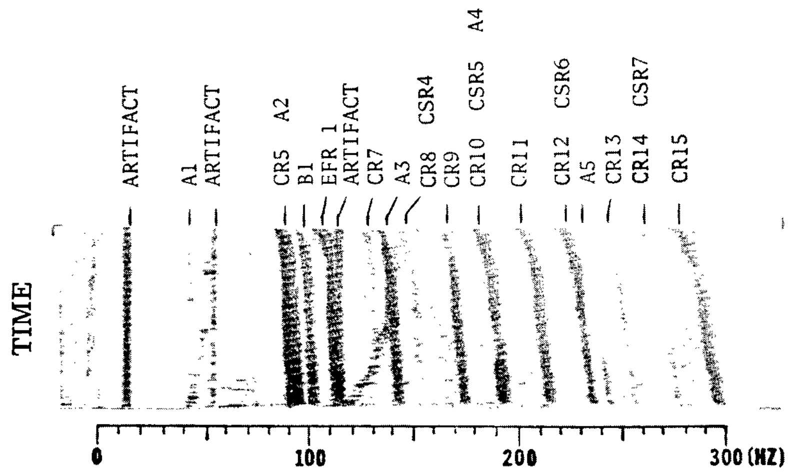
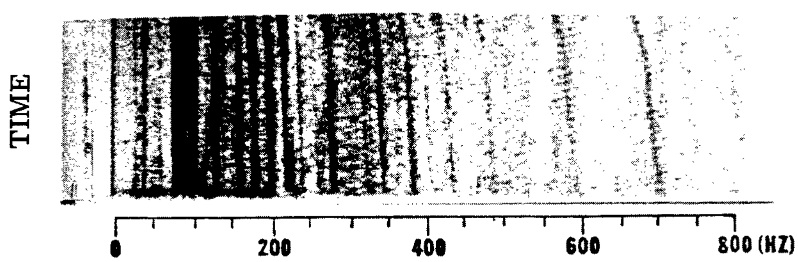


Figure 4-12. Spectrogram, Broadband and Narrow Band, for a 5-Ton Diesel Truck at 20 mph, 4th Gear





This signature is from a Dodge 3/4-ton Power Wagon with a weapons carrier body and a 4-cycle, 6-cylinder, gasoline engine operating at 2280 rpm. Signature is not typical as vehicle is stationary. The CFR is 19.0, CSR is 38.0, and EFR is 114.0. The A-line is probably from the water pump and the B-line is probably from the generator. A-SR is 46.3 at 2780 rpm, thus A-SR/Eng-CSR is 1.22/1. B-SR is 103.0 at 6180 rpm, thus B-SR/Eng-CSR is 2.73/1. No body resonance reinforcement effect is seen, as the signature appears exceptionally uniform across the spectrum.

Figure 4-13. Truck Classification by Spectral Analysis

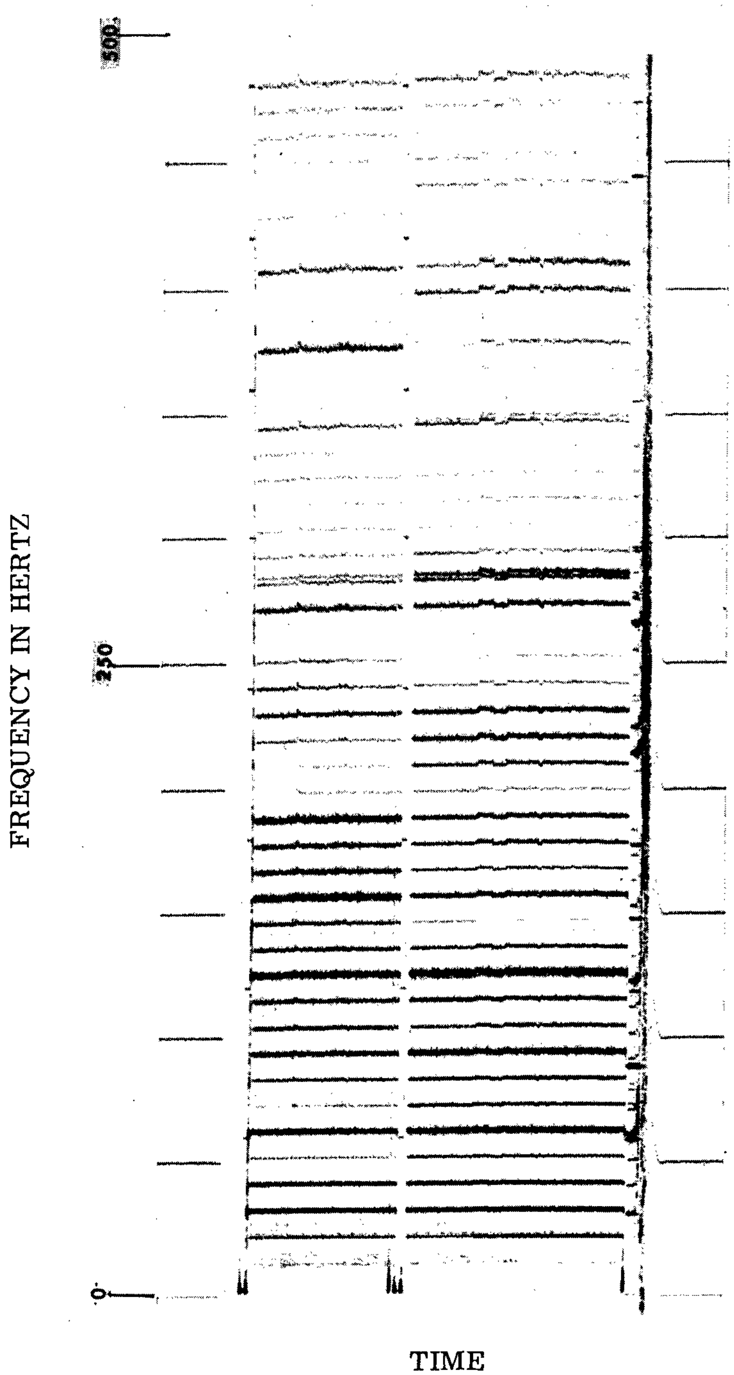


Figure 4-14. Engine Idle Spectrum, Large Reciprocating Engine

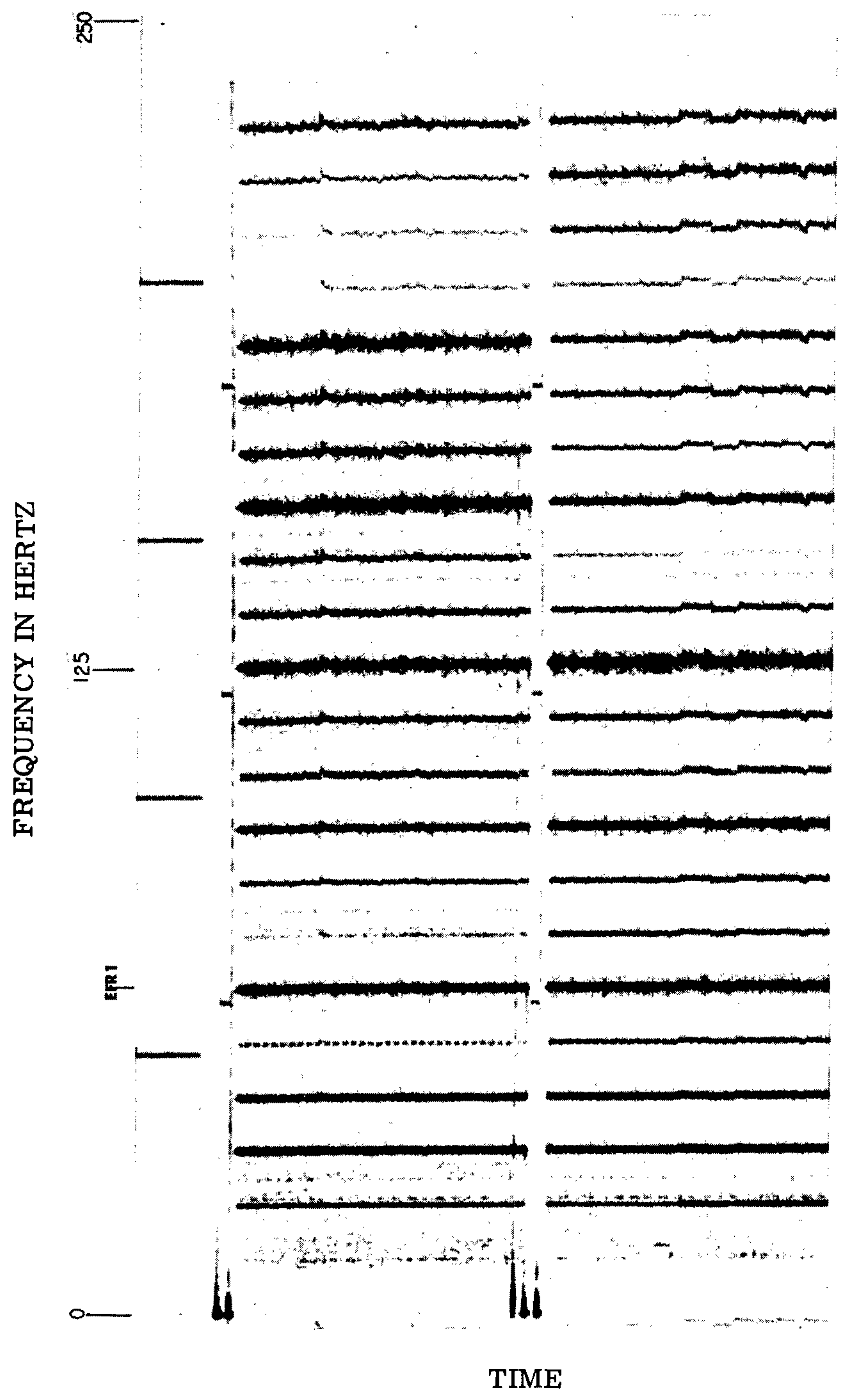


Figure 4-15. Engine Idle Spectrum, Large Reciprocating Engine, Portion of Figure 4-14 Expanded for Greater Detail

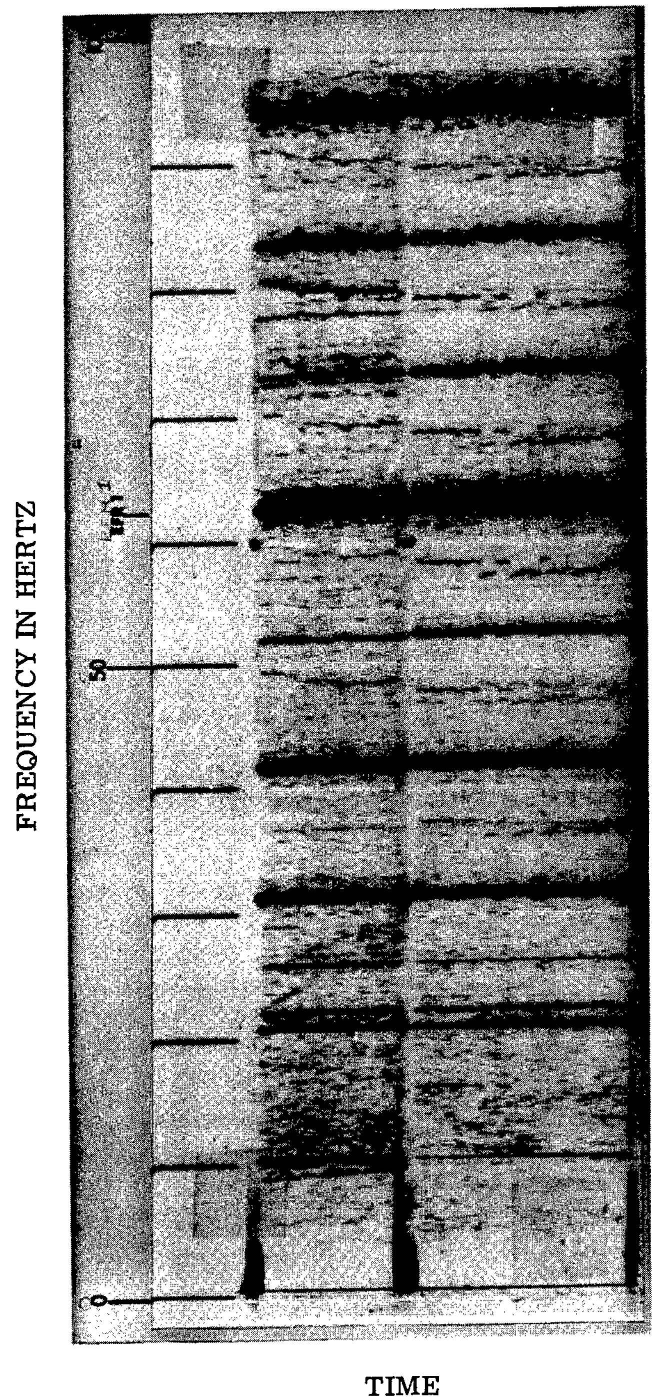


Figure 4-16. Engine Idle Spectrum, Large Reciprocating Engine, Portion of Figure 4-15 Further Expanded

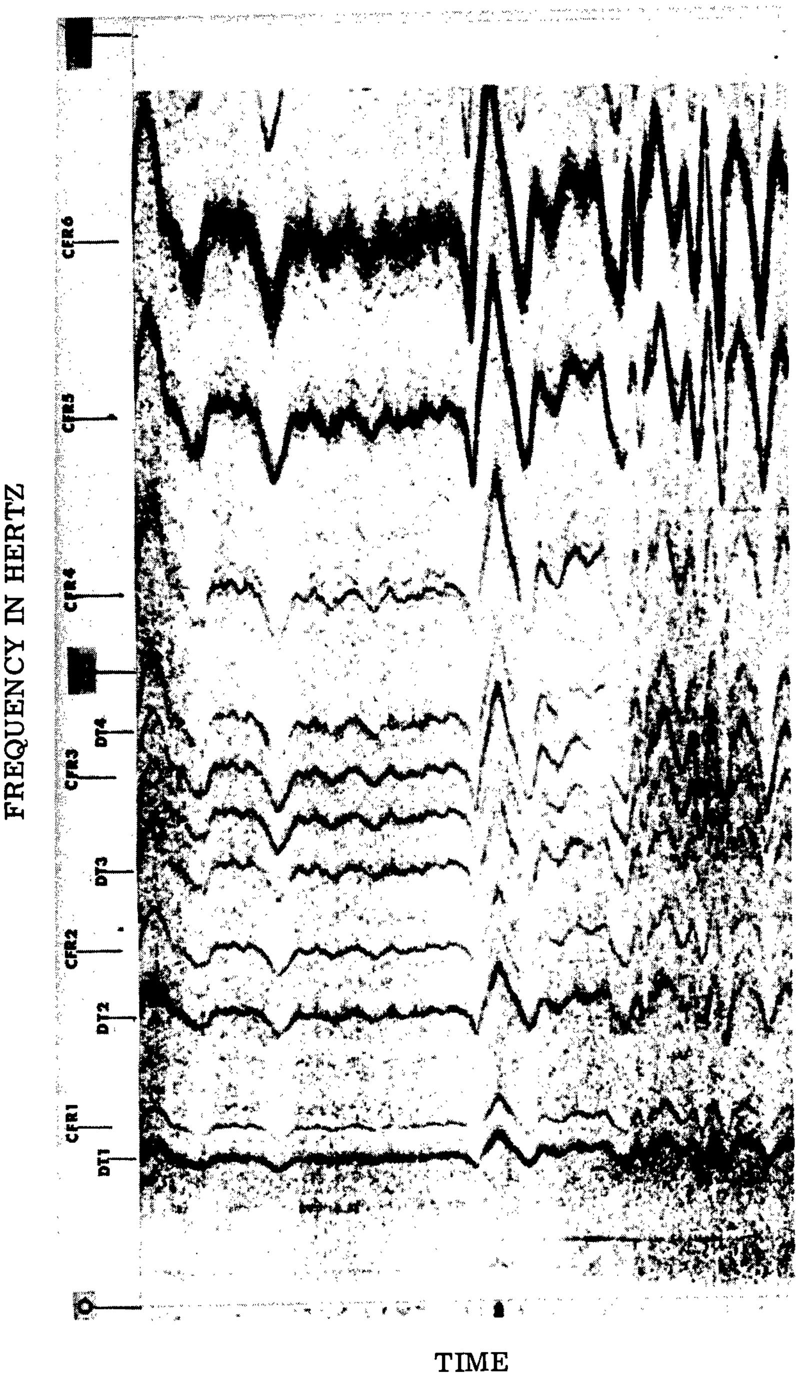


Figure 4-17. Large Reciprocating Engine, 2nd Gear Spectrum

4.3.C-4.3.D

Drive Train and Accessory Lines (DT) Components are useful for classification purposes.

Accessory lines are not necessarily harmonically related to other engine lines.

CFR and DT lines can usually be placed in a harmonic set. (See Engine Idle Signatures, Figures 4-14 to 4-16.)

Table 4-5 (p. 4-23) gives conventional calculations for engine frequencies. However, these cannot be relied upon to give the exhaust frequencies in engines whose dimensions are such that the physical spacing between successively firing cylinders approaches the distance travelled by an exhaust pulse between firings.[62] In these cases there are several frequencies other than the conventionally calculated one, and the latter will be absent.

Spectrograms indicating EFR, CR, CSR harmonics for a Dodge 3/4-ton Power Wagon (4-cycle, 6-cylinder gasoline engine operating at 2280 rpm) are illustrated in Figure 4-13.

Engine Idle spectrograms for a large reciprocating engine (Figures 4-14, 15 and 16) illustrate the large number of lines that may be observed. Figures 4-17 and 4-18 show the result of transient effects when the engine is operated successively in 2nd and 4th gear and contrasts with the idle condition above. The sharpness of lines in these spectrograms attests to the discreteness of engine frequencies when observed with suitable equipment.

D. Aircraft

In considering aircraft sounds one would like to have the following field-collected data:

1. Spectral characteristics of aircraft as acoustic sources.
2. Acoustic levels generated by aircraft as a function of their distance from the sensor.
3. Influence of altitude, terrain, aspect, clouds (atmospherics in general) as well as aircraft type and mode of operation.
4. Spectral characteristics of masking phenomena such as vehicles, fauna and various other natural or man-made sounds.
5. Time domain features of masking phenomena.

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[62] W. B. Diboll, "The Measurement and Prediction of Exhaust Frequencies in Large Internal Combustion Engines," ASME 72-DGP-2, American Society of Mechanical Engineers.

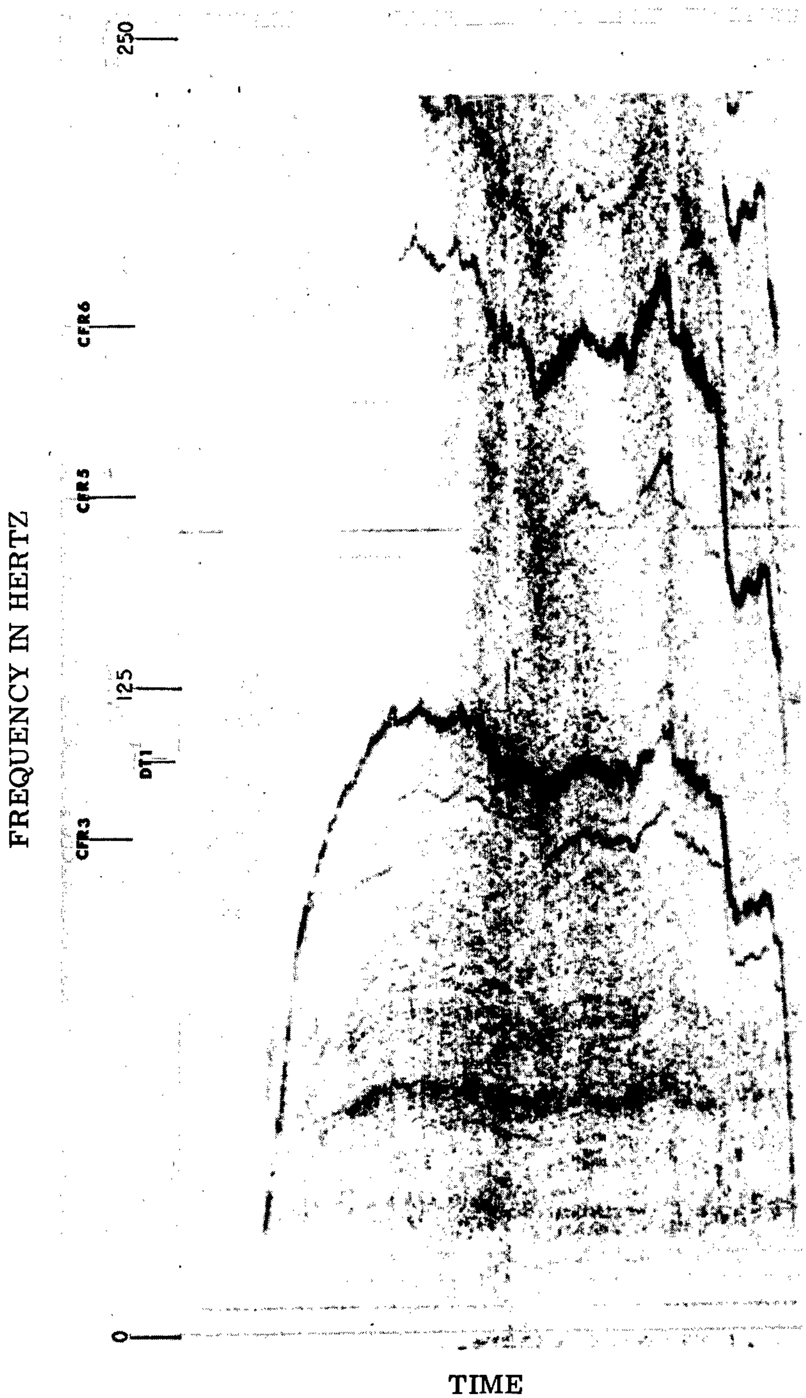

Figure 4-18. Large Reciprocating Engine, 4th Gear Spectrum

4.3.D

Since many of these factors are interrelated, i.e., influence each other, and constitute a large number of variables, a rather extensive measurement program would be required to present the total picture. Moreover, there are no completely adequate means of determining and describing such things as terrain characteristics and atmospherics. Therefore a degree of generalization inevitably enters.

In the data which follow some of the listed factors are ignored. For example, aspect effects become merged with changes in distance during overflight experiments. An aircraft at 10,000 ft. altitude is still 2 miles distant when it is directly overhead, but its aspect in this condition is quite different than when 2 land miles away.

General evaluations or comparisons of the sound level gradients (with time) for different types of aircraft must be made with care. It is important to note in this regard that a particular jet aircraft may be especially fast and hence produce different gradients for this reason. The large differences in speed during various stages of flight will of course influence its characteristic as a sound source.[63]

The investigation of aircraft noise goes back to the National Advisory Committee for Aeronautics (NACA), NASA's predecessor. NACA publications on aircraft engine mufflers appeared as early as 1916.[64] The problem of civil aircraft noise and its elimination was similarly reported in 1928. NASA, formed in 1958, has become a major source of information on the mechanisms of noise generation in jets. The fundamental work of Lighthill[65] in England in the 50's provided the basis for the analysis of these phenomena. Another area of NASA investigations of interest to us concerns turbojet aircraft engine compressor noise. To gain entry into the extensive literature on the theory and characteristics of these noise

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[63] See pp. 2-4 in footnote 46, p. 4-6.

[64] Diederichs, H. and G. B. Upton, "Mufflers for Aeronautic Engines," National Advisory Committee on Aeronautics, NACA Report No. 10, 1916.

[65] Lighthill, M. J. "On Sound Generated Aerodynamically," Proc. Royal Soc. (London), 1952, 1959.

sources reference is made to a 1970 NASA technology utilization publication.[66]

The aircraft acoustic data contained in the present Navy (NADC) publication are less concerned with the mechanisms of noise production and ways of abatement. Rather the contribution of recent NADC work lies in reporting practical characteristics of the measured sounds and ways in which they may be identified selectively.

It is to be expected that among all the many reports giving spectral data on aircraft sounds there must be some variation in the data presented. One cannot say therefore that a definite spectral characteristic, such as -6 dB per octave, etc., exists. More realistically one displays typical data for a variety of craft in a number of situations.

The -6 dB per octave slope is noted on the jet characteristic in Figure 4-19. However, the data given in Table 4-6 can be shown to deviate from this form in various ways depending on conditions of flight. We see spectral characteristics in general that vary between -2.5 and -10 dB per octave. With all of these factors in mind we present the following data:

Table 4-6. Summary of Aircraft Signatures

TYPE OF AIRCRAFT	AIRCRAFT SIGNATURES*
Jet	Impulsive in nature, most of energy below 300 Hz. -6 dB/octave with increasing frequency (Figure 4-19). Smooth analysis spectrum obtained with 20 Hz bandwidth.
Prop	Line spectra apparent when analysis bandwidth is 2 Hz, smooth spectrum when analysis bandwidth is 20 Hz. Spectral peaks below 200 Hz.

*Signal energy primarily below 300 Hz

[66]See pp. 52-65 in footnote 47, p. 4-8.

4.3.D

The spectra of jet aircraft are fairly smooth below 1000 Hz and contain no outstanding line components (Figure 4-20). Signature Contour Mapping has been employed to present the same data in more compact form (Figure 4-21).

The spectra of piston aircraft are smooth over the 50 to 500 Hz band, but there is evidence of one or more outstanding line components (Figure 4-22)shown by broad peaks in this spectrogram. The peaks are not better resolved here because of the one-third-octave analysis limitation. These analog data were analyzed in real time (Table 4-7 , Type 2).

Figure 4-7. Typical Response Characteristics for the Ceramic Microphones (a) 1-in., (b) 1/2-in.

TYPE TEST CONDITION	DESCRIPTION
1	Time Domain presentation of broad-band data via Honeywell "Visicorder." Spectral analysis, 100 Hz bandwidth to 5 kHz.
2	A bank of one-third octave filters covering the 8 Hz to 1600 Hz frequency range.

A comparison of jet and piston aircraft spectra in Figures 4-20 and 4-22 leads one to conclude that a logical discrimination decision can be based upon the difference in approach and recession spectrum levels. It will be noted that in the 500 Hz region the jet approach levels are 40 dB higher than the recession levels, while the piston aircraft's corresponding levels are low and and about equal. In fact, the relationships of the entire approach and recession curves are reversed for piston and jet aircraft.

A digital program for performing a Fast Fourier Transform was devised so that a one-third-octave spectrum could be computed from the same recorded data. Results for a jet are shown typically in Figure 4-23, and for piston aircraft in Figure 4-24.

Unfortunately the scales used do not make matching between analog and digital analyses easy. To obtain agreement among the several forms of data presentation, a correction is required when comparing the analog spectrograms, which are not amplitude normalized to an effective 1 Hz bandwidth, with the digital computation

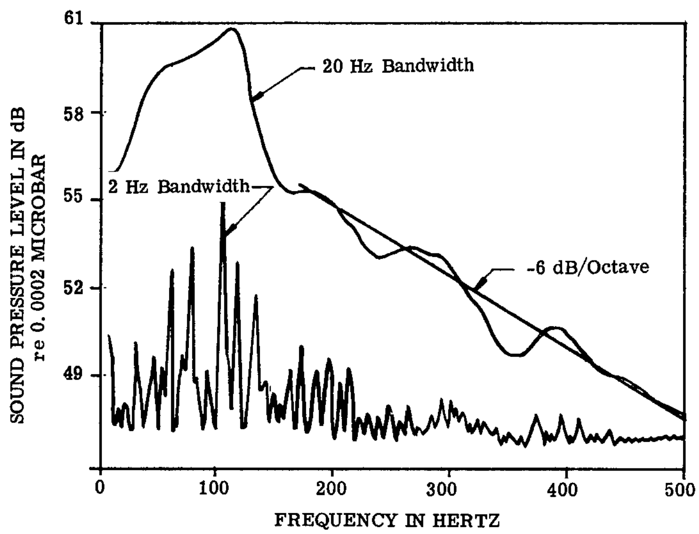

Figure 4-19. Jet Aircraft Characteristics

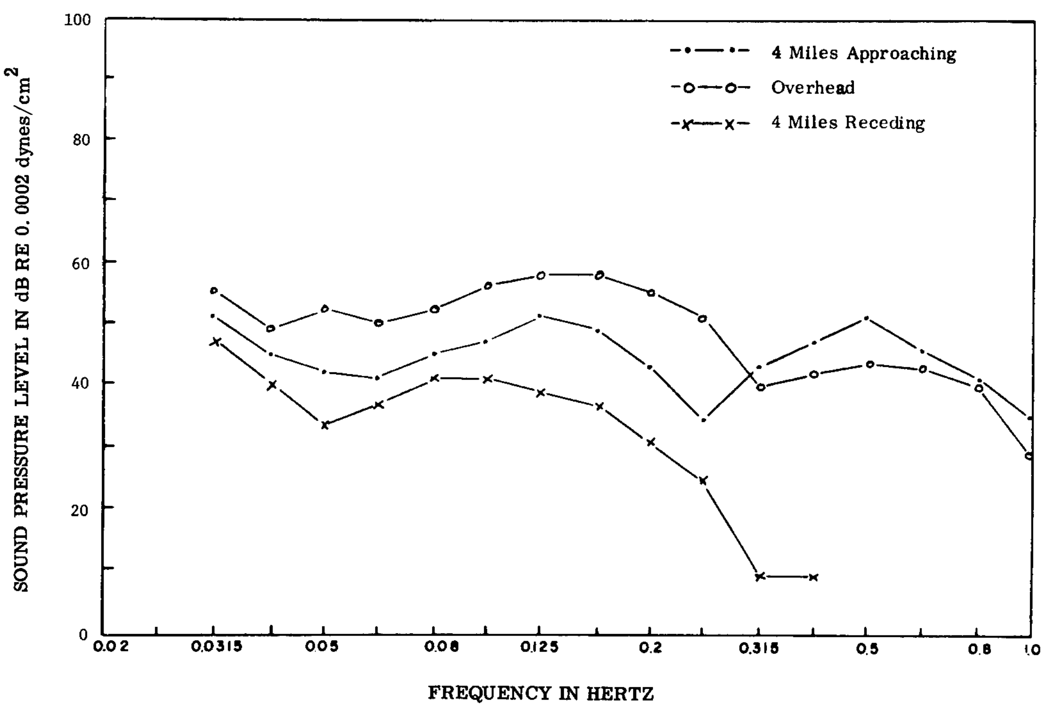

Figure 4-20. Jet Aircraft at 10,000-ft Altitude

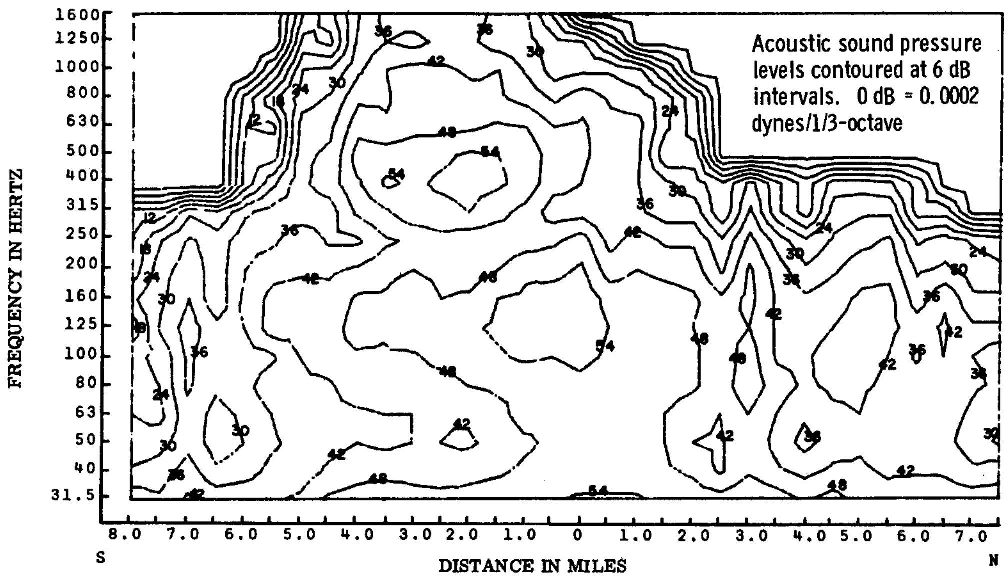

Figure 4-21. Jet Aircraft Signature Contour Map 10,000 ft Overhead

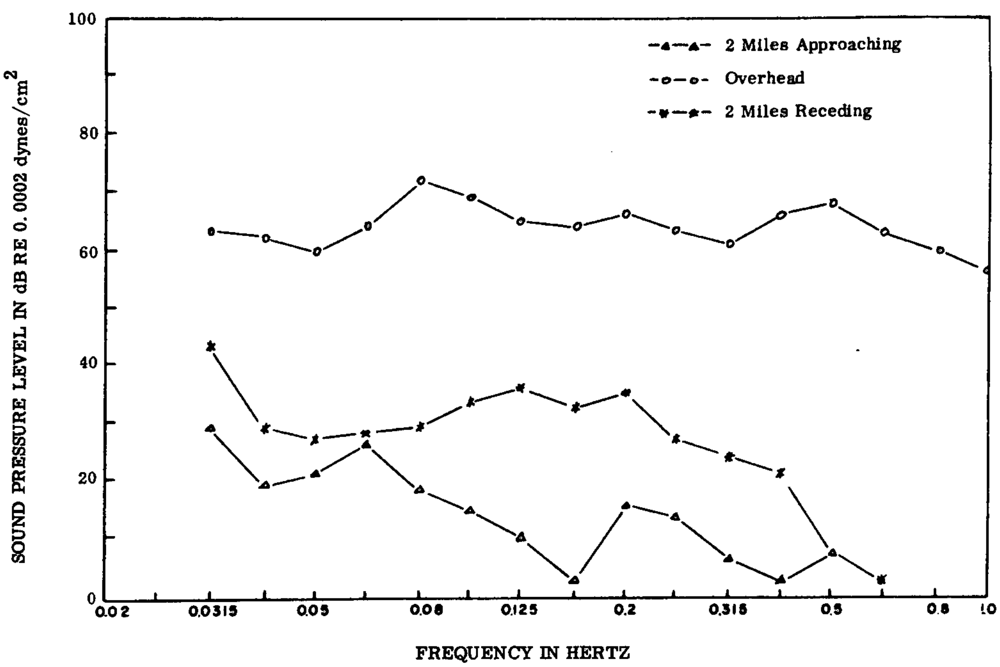

Figure 4-22. Piston Aircraft at 1000-ft Altitude

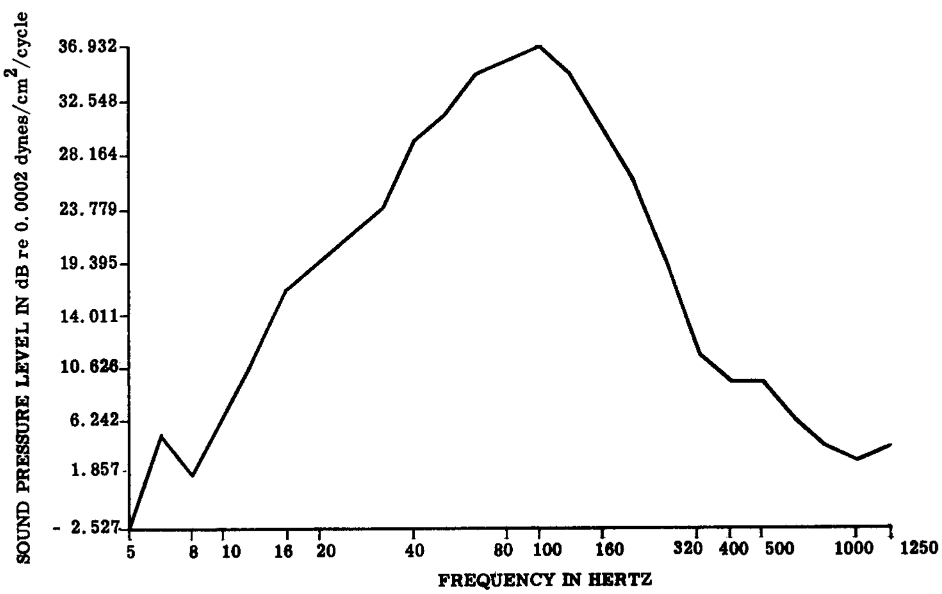

Figure 4-23. Jet Aircraft at 10,000-ft Altitude, 1/3-Octave Analysis Digitally Computed Fast-Fourier Transform, Normalized to 1 Hz Effective Bandwidth

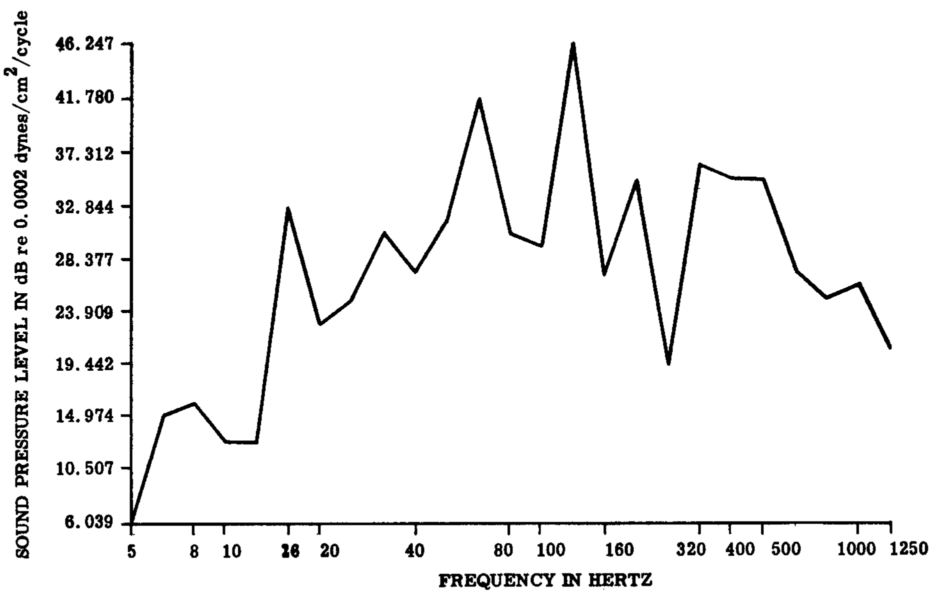

Figure 4-24. Piston Aircraft at 5000-ft Altitude, 1/3-Octave Analysis Digitally Computed Fast Fourier Transform, Normalized to 1 Hz Effective Bandwidth

4.3.D

(Fast Fourier Transform method) which is normalized (see p. 2-40). A correction must be added to the latter as follows:

The correction, $c = 10 \log (f_a - f_b)$,[67] where f_a is the lower cutoff frequency and f_b is the upper cutoff frequency, must be added. For one-third-octave analysis, $f_a - f_b$ equals 23% of the center frequency of the filter. (See Figure 4-2 for illustration of this correction applied to one-third-octave data).

A typical time spectrogram for a small jet at 1000 ft shows the distribution of energy slopes at approximately 2.5 dB per octave (Figure 4-25). The characteristic moiré pattern is produced by a source whose energy is distributed continuously over the frequency range and the multipath propagation geometry between the source and the microphone. Certain bands of frequencies experience degrees of reinforcement, and others destructive interference. It must be observed that the pattern is a product of the propagation geometry and not the source.

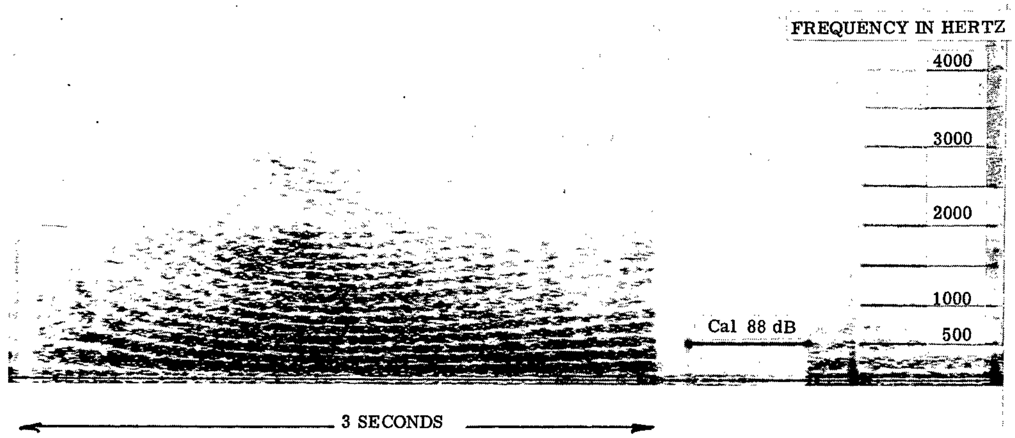

Figure 4-25. Jet Aircraft Signature Showing Effect of Multipath Propagation

The influence of altitude on the acoustic levels generated by an aircraft as a function of distance from the sensor in surface miles is illustrated by the two plots in Figure 4-26. Here we see that the peak level and especially the gradients are affected.

[67]See pp. 563-564 in footnote 35.

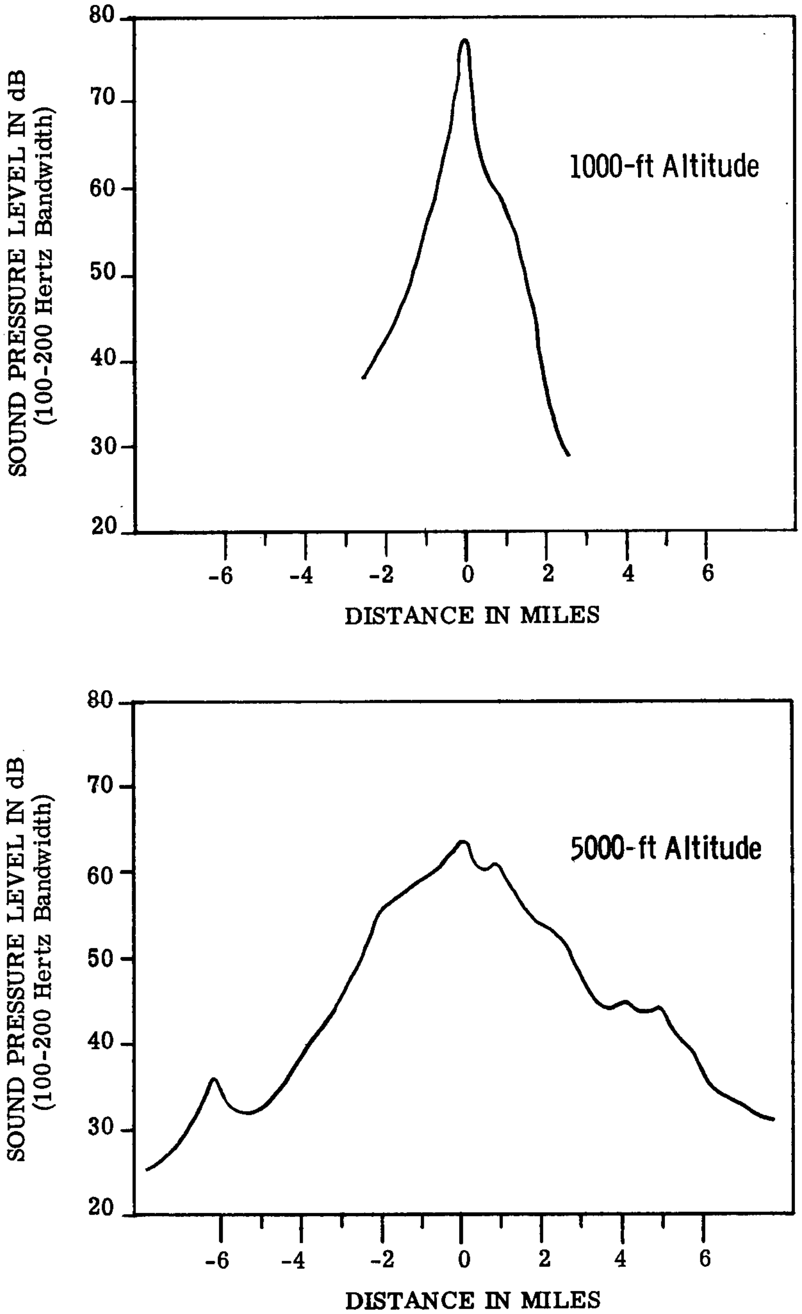

Figure 4-26. Piston Aircraft Overflight

E. Personnel

A number of common personnel sounds are considered. These include walking (footsteps), talking, coughing, outdoor wood chopping. Footsteps out of doors, except on pavement or on snapping branches, etc., are generally faint and hard to distinguish at ranges of more than a few yards. Speech, on the other hand, follows the same type of characteristic as the commonly used SIL (Speech Intelligibility Level) curves which depend upon ambient noise level. Some possible levels are given below.

Personnel Signatures[68,69]

- Footsteps at 10 feet (detection range), outdoor.
  - Signal Peak: 36 dB rel. to 0.0002 μbar (outdoor ambient noise 21 dB)
  - Fluctuations $\pm$ 1 dB. Depending upon time constants of recording system used.
  - Most of energy in 1 to 3 kHz region
  - Terrain: muddy, rough and broken surface.
  - Test conditions: Table 4-8, Type 1
- Talking, outdoor. Southeast Asian voices 1 octave higher than American. 100 ft. (detection range). 42 dB peak, main energy below 1 kHz.
  - Test conditions: Table 4-8, Type 2
  - 60 dB (classical male voice) at 1 ft.
  - Talking, Speech Intelligibility Level (SIL)
- Chopping, outdoor at 100 ft (detection range)
  - Using "Energy Detection Systems," adaptive filter and correlation could be better.
  - Cutting 6-in. diameter trees.
  - 41 dB peak to 4 kHz at 96 ft.
  - 1 to 3 kHz (classical wood chop)
  - 39 dB P.S.L., 1 Hz bandwidth, at 1 ft.
  - S/N 13 dB at 100 ft
  - Test condition: Table 4-7, Type 2

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[68] NADC, "Preliminary Report of POTLID Personnel Test," 22 August 1968, pp. 8, 9.

[69] See "Personnel Detection," p. 24 in footnote 59, p. 4-21.

F. Impulse

The ignition and firing of engines, detonations, ordnance explosions of all sizes and kinds constitute impulsive sound. Under Natural Sounds we have mentioned nearby thunder (Table 4-2) and the phonations of certain fauna.[70] A 1.5-gram small explosive may be taken as a good impulse example. On rock it produces a peak SPL of approximately 95 dB re 0.0002 μbar at 100 ft, the spectral peak below 1 kHz. The harmonic structure falls from 90 dB at 1 kHz with a declination of approximately 2 dB per octave after the initial fall rate of 10 dB at 1 kHz.

Test conditions for the data given above were Type 4, Table 4-8.

Table 4-8. Test Method and Equipment

TYPE TEST CONDITIONS	DESCRIPTION
1	GR P5 microphone (3 dB bandwidth 5 Hz to 8 kHz -10 kHz), system upper frequency limitation. High, 20 ft, and low, 2 ft, placement at three locations, 10, 20 and 40 ft from the trail. Microphones, 300 ft from recording site, utilized P40 preamp. set on X10. Recorders, Lockheed 417, used 1/2-in. tape at 3-3/4 ips, 1 V rms input for full record level, direct record mode (3 dB bandwidth 100 Hz to 12.5 kHz), limited system lower end frequency response. SLM, GR 1551C, monitored with earphones and scopes as was recorder output. GR 1562 calibrator used as well as a square wave input to 1551C each reel.
2	Same as 1 but microphone placement at 16, 32, and 96 ft
3	Recorder, Lockheed 411C, 1/2-in. reel, 7-1/2 ips, FM, Analyzed 2 Hz and 20 Hz bandwidth.
4	2203 B & K Sound Level Meter. B & K microphone. Recorder, Kudelski, 1/4-in. full track; 15 ips direct record. Analysis by GASL , analysis bandwidth 100 Hz, sample size 0.1 sec.
5	Recorder, Lockheed 411C, 7-1/2 ips, direct. Analysis bandwidth 100 Hz, input filter 5 kHz bandwidth.

[70] See p. 5 in footnote 51, p. 4-10.

4.3.F-4.3.G

At 500 ft a 1.5-gram explosive produced a spectral peak of 75 dB below 1 kHz, the characteristic falling at 10 dB per octave (Figure 4-27). Test conditions here were Type 5, Table 4-8. A number of analyses shows a secondary spectral peak at 2 kHz, approximately half the height of the main one. The recordings which display a double peak are usually taken at 100 ft range. The calculated impulse source level at 1 ft is approximately 140 dB (0.0002 dyne/cm^2,200 - 500 Hz).

The spectral decay rate varies markedly when firings are shifted from open ground into a woods. For example, in the open the spectrum decays at 7 dB/octave above 2 kHz, while in the woods it is 14 dB/octave.[71] A listener's subjective impression of the tone of such sounds surely would be influenced by this effect, all of which reemphasizes the fact that the surroundings at the time the sound characteristics were recorded may have an important effect upon what we say they are.

Impulse sounds generally are detected by equipment designed for that purpose if they rise above the background by at least 12 dB and the signal rate of increase is 16 dB/sec or more.[72]

G. Boats (Air Path Over Water)

Motorized Boats:[73]

The boats used in these tests were wooden, round-bottomed, approximately 16.5 ft in length.

The engine was a one-cylinder, one-cycle, Clinton rated at 4-1/2 horsepower running at 1430 rpm under load with full throttle. Essentially inboard installation with long direct-coupled shaft characterized the mechanical arrangement.

Ranges studied varied from 15 ft (CPA) to 400 ft.

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[71]NADC-SY-6703, "Analysis of Acoustical Data from Tests Performed on 27 November, 1966," 14 March, 1967, p. B-3.

[72]See p. 24 in footnote 59, p. 4-21.

[73]NADC-AWG-31, 10 Jan. 1969, pp. 24-29. (Spectrograms given in Appendix consists of CBS Laboratories' Rpt. No. 5025-DR-027, pp. 34-61)

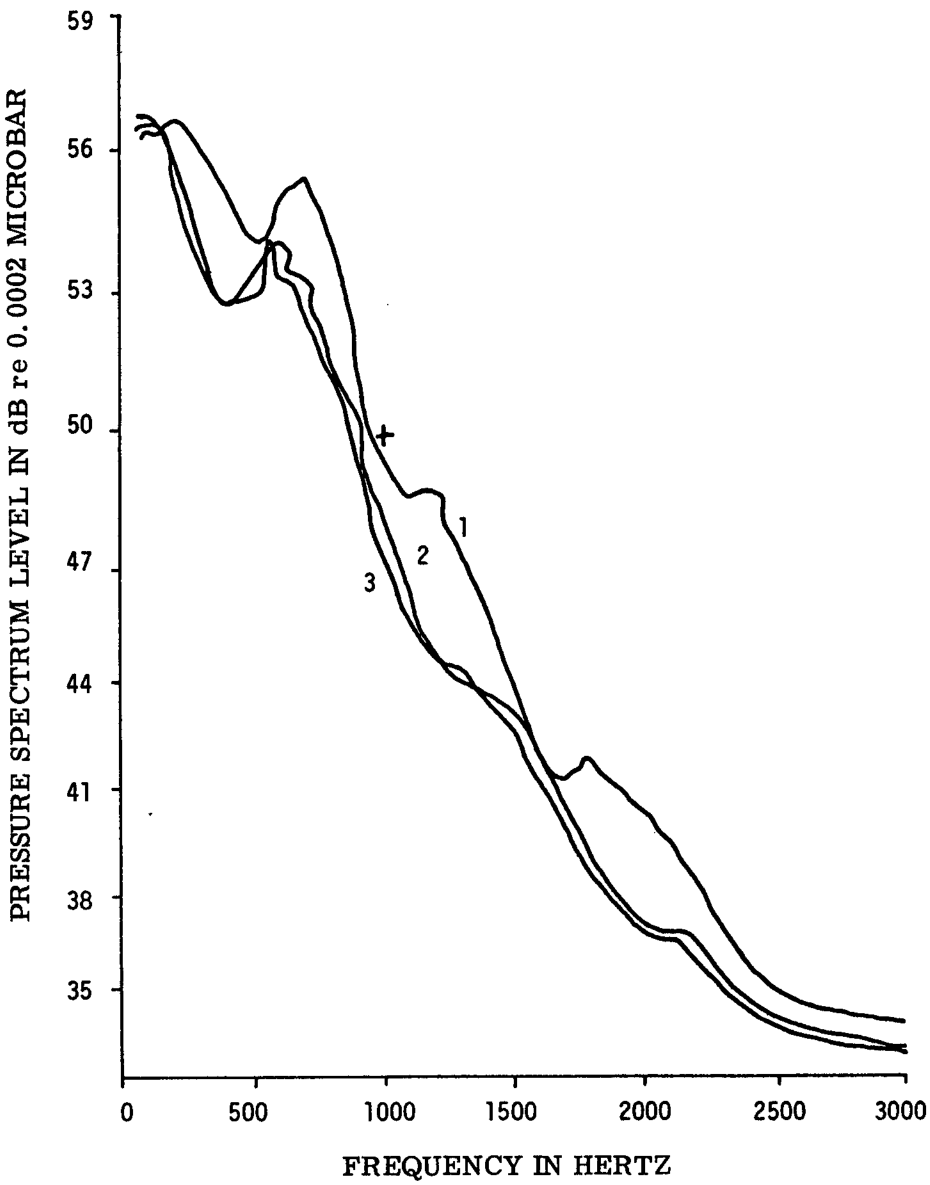

Figure 4-27. Spectrum Analysis for 1.5-Gram Explosive Shots at 500 Feet, Range Site 1, East Leg, 100-Hz Analysis Bandwidth

4.3.G-4.3.H

The spectrum (25 to 2500 Hz) was smooth at 45 dB from 40 to 500 Hz, then -10 dB per octave to 1 kHz. Some data indicates a 100 Hz peak, then 3 dB per octave to 1.5 kHz. This data applies to an air microphone at ranges 15 to 400 ft and 1/10 octave SPL analysis. Under these conditions levels of 70 dB at the closest point of approach, CPA, (15 ft) are reached.

The microphone data shows strong line frequency structure based upon the engine reciprocation rate of 12.2 Hz for 1430 rpm. This structure is also somewhat apparent in the hydrophone data.

H. Boats (Underwater Sound)

Motorized Boats:[74]

The boats used in these tests were similar to those described above.

Shallow hydrophone data analyzed on the basis of 1/10-octave shows peaks at 40 Hz which are -20 dB re 1 dyne/cm^2, then -10 dB per octave from 40 Hz to 400 Hz. However, the hydrophone optimum energy band for these boats is in the 2000 Hz to 9000 Hz region because of reduced background noise in this part of the spectrum. The sound appears to be caused by cavitation near the propeller.

The peak pressures reached at the CPA in the 2000 Hz to 9000 Hz band are 110 and 140 dynes/sq cm for deep and shallow hydrophones, respectively.

A secondary, low-frequency region, probably associated with the propeller rate, also appears.

Electric Discharge[75]

This method of generating strong acoustic pulses in water has been studied and shown feasible although not efficient in seawater owing to the latter's high conductivity. Electric capacitors of 15 μF and 20 kV rating have been used in a system having a maximum energy capability of 9000 J. Acoustic pressure in fresh water was 156 PSI peak (140.6 dB re 1 degree/cm^2) at one yard for an initial energy of 1700 J. In salt water an initial large current reduces the available capacitor voltage, hence peak discharge current and pressure.

[74] See pp. 42, 43, 76 in footnote 73, p. 4-42.

[75] Fairchild Hiller, "Studies on an Electric Discharge Underwater Sound Source," Rpt. No. PCD-TR-68-16.

Large Ships

The periodic pulsation sounds of large ship propellers are caused by cavitation. Frequently the beat is pulsating in accordance with the number of blades, and occasionally one blade will be noiser than the others, thus producing an accented signature.[76] In such cases the shaft rotation rate and number of blades can be determined.

The spectra of ship sounds in general are continuous. This broad distribution of energy decreases in sound level at 6 dB per octave with increasing frequency. Most of the audible output is below 2000 Hz. There is some pure tone content, and a time pattern of intensity which may be subject to interpretation as to origin or cause.[77]

I. Aerodynamic Noise

Broadband noise generated by various forms of gas turbulence results from the operation of modern aircraft. Jet-generated turbulence is evident everywhere. Fundamental mechanisms[78] of sound generation by this type of source have been completely disclosed in the literature related to aeronautics and astronautics, industrial pneumatic systems, etc., and measured characteristics are given.

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[76] See Vol. 7 in footnote 41, p. 3-16.

[77] L. W. Camp, "Underwater Acoustics," Interscience, John Wiley & Sons, N. Y., 1970.

[78] See Vol. 211A, 1952 and Vol. 222A, 1959 in footnote 65, p. 4-32.

4.4 TOPICAL SUMMARY OF USEFUL SOUND DATA

Useful sound (natural and man-made) data is obtainable from NADC documentation according to the listings appearing on this page and the next.

NATURAL	
SUBJECTS	SIGNATURE DATA GIVEN IN
Laos Ambient	AWG-32
Thunder	Howard, J. R. and S. Krieg, Letter Rpt. No. 22, 12 June 1967
Wind	5025-MR-007 5025-DR-029 U. of Mich. 8510-4-F
Rain	5025-DR-010 5025-DR-011 5025-DR-012 5025-DR-013 5025-DR-015 5025-DR-016 5025-DR-017 NADC-SY-6703, 4, 5, 6
Insects	8510-4-F
Bird Whistle	5025-DR-017 5025-DR-022
Running Stream	U. of Mich. 8510-4-F
Ocean Ambient	NADC-SD-7143
Fauna	Howard, J. R. and S. Krieg, Summary Rpt. of Acoustical Data for Phase II System, 10 Jan. 1967

MAN-MADE	
SUBJECTS	SIGNATURE DATA GIVEN IN
Vehicles	AWG-31 5025-DR-027 NADC-SD-7179 NADC-SD-7191
Trucks	AWG-32 AWG-33 AWG-34 AWG-37 NADC-SY-6715 NADC-SY-6803 U. of Mich. 1099-5-F 5025-DR-008 5025-DR-009 5025-DR-010 5025-DR-013 5025-DR-016 5025-DR-026 NADC-72058-SD
Explosions	AWG-35 NADC-SY-6703 NADC-SY-6704 NADC-SY-6705 NADC-SY-6706 U. of Mich. 8510-1-P U. of Mich. 8510-4-F Howard, J. R., S. Krieg, Summary Rpt. of Acoustic Data for Phase II System, NADC, 10 Jan. 1967 5025-DR-012 5025-RT-008 5025-RT-009 5025-RT-010
Aircraft	NADC-SY-6803 NADC-SY-6715 NADC-72058-SD 5025-DR-026
Motorized Boats	AWG-31 SSG-36 5025-DR-027 5025-DR-028
Rocket Mortars	AFCRL-68-0457 SSG-38
Pistol	5025-DR-012
Shots	5025-DR-017

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APPENDIXES

WRITTEN FIELD PROCEDURES

APPENDIX A

FIELD CALIBRATION PROCEDURE

1. Set up equipment and make the following announcement: "This is a system calibration for test ______________. This tape is being recorded at ______ ______________________________ Site _________. Today's date is ___________. Personnel at this site are ____ ____.

This recording is being made on a Lockheed _______ machine operating at ______ ips. Track 1 is recording the voice commentary in the direct mode. Track 2 is recording the data gathering microphone(DGM)in the direct mode. Track 3 is recording the IRIG-B time code in the direct mode. Track 4 is recording the hardwired item in the direct mode. The location of the microphones is shown on the accompanying sketch." (At this point describe site, terrain, foliage, local anomalies, etc., and describe weather conditions).

2. Stop recorder. Short channels 2,3, and 4 at the recorder input terminals with BNC shorting caps. Record for at least 30 seconds and annotate: "This signal is a measure of recorder noise made with the recorder inputs shorted at the recorder terminals."

3. Stop recorder. Remove shorting caps and reconnect system. Short data lines (Tracks 2 and 4) at the microphone ends using shorted Pigmy connectors.

4. Set both SLM's at 110 dB. Record for at least 30 seconds. Annotate: "This is a recording made with the cables shorted at the microphone end. The SLM's are set at _______ dB ."

5. Repeat 4 for SLM settings of 100 dB, 90 dB, 80 dB, 70 dB, 60 dB, 50 dB, 40 dB.

6. Connect the DGM (data gathering microphone). Start recording and announce into the DGM "I am speaking directly into the DGM, serial number ___________, at Site _________." Annotate on commentary track: "A signal is now being recorded on Track 2 in the direct mode. This signal is coming from the DGM, serial number ___________. The serial number of the associated SLM is _________."

7. Disconnect the DGM.

8. Connect the hardwired item. Start recording and announce into the hardwired item: "I am speaking directly into a hardwired item, serial number _________ at Site _________." Annotate on commentary track: "A signal is now being recorded on Track 4 in the direct mode. This signal is coming from the hardwired item, serial number _________. The serial number of the associated SLM is _________."

9. Reconnect the DGM. Set SLM at 110 dB. Annotate: "The signals which will be recorded come from a General Radio 1562A Sound Level Calibrator. In each case the level is 114 dB re 0.0002 dynes/cm^2. The signals serve to calibrate the DGM recorded on Track 2. The SLM is set at 110 dB." Record at least 30 seconds of each frequency on the calibrator annotating each as follows: "The present signal is _____________Hz."

10. Disconnect the DGM microphone cable from the SLM. Set SLM at 110 dB. Insert CBS square wave calibrator into SLM. Set calibrator to 250 Hz and depress button. Annotate: "I am recording a 250 Hz square wave on Track 2. The

level of the 250 Hz fundamental is ________ volts rms. The SLM is set at 110 dB. This signal will serve to calibrate future reels of tape taken at this site."

11. Set up the trumpet near the hardwired item. Hold a calibrated sound level meter with P5 microphone near item (within 3" of microphone). Take care not to influence the sound field with your body, i.e. do not shield either microphone or stand directly in back of the microphones. It is best to stand to the side so that you neither block the free passage of sound nor serve to reflect it. Set up a 500 Hz sound field using the modified Bogen power amplifier and driver and trumpet. The level at the buoy should be well above ambient noise. Record at least one minute. Annotate: "The signal recorded on Track 4 is from the hardwired item. The sound pressure level at the item measures ________ dB$\pm$ ________ dB. The frequency is approximately 500 Hz. The SLM is set at ________ dB."

12. Disconnect the hardwired item cable from the SLM. Set SLM at 110 dB. Insert CBS square wave calibrator into SLM. Set calibrator to 250 Hz, and depress button. Annotate: "I am recording a 250 Hz square wave on Track 4. The level of the 250 Hz fundamental is ________ volts rms. The SLM is set at 110 dB. This signal will serve to calibrate future reels of tape taken at this site."

13. Annotate: "This completes the basic calibration of the data gathering system at Site ________. This calibration tape is to be used to establish

the relationship between recorded signal level and acoustic sound pressure levels on future reels of tape taken at this site during this test. The method of correlation is as follows: In reducing this tape, the output level of the reproducer is noted for the two acoustic calibration sections, i.e. the 114 dB SPL section recorded on the DGM using the General Radio Acoustic Calibrator, and the 500 Hz acoustic signal generated by the trumpet and recorded by the hardwired item. Immediately following each of these sections is a section of 250 Hz square wave. In reproduction, the level of the 250 Hz fundamental is established with a 1/10-octave filter. This level is then correlated to an equivalent acoustic sound pressure level. Every future reel of tape will have a 250 Hz calibration square wave recorded on it. The level of the fundamental of this signal will then establish an equivalent known acoustic sound pressure level."

APPENDIX B

FIELD TEST OPERATION PROCEDURE

1. Annotate: "This is test __________. The purpose is to record basic acoustic data on airfield activity. This is reel __________ recorded at Site ______________________________.
This site is located approximately __________ feet from the south runway.
Today's date is __________________. The time is ____________ CDT.
The recording speed is ____________ ips on a Lockheed ____________ machine. Track 1 is recording the voice commentary in the direct mode. Track 2 is recording the data gathering microphone (DGM) in the direct mode. Track 3 is recording IRIG-B time code in the direct mode. Track 4 is recording the hardwired item in the direct mode. The location of the microphones is shown in the accompanying sketch. The microphones are suspended ________ feet above the ground."

2. Set Track 2 SLM to 110 dB. Disconnect cable from the DGM and insert square wave calibrator. Set calibrator to 250 Hz and depress button. Record at least 15 seconds and annotate: "I am recording a 250 Hz square wave on Track 2. The level of the 250 Hz fundamental is __________volts rms. The SLM is set at 110 dB. In conjunction with the calibration tape, the signal can be used to establish acoustic sound pressure levels."

3. Reconnect cable from the DGM.

4. Set Track 4 SLM to 110 dB. Disconnect cable and insert square wave calibrator. Set calibrator to 250 Hz, and depress button. Record at least 15 seconds and annotate: "I am recording a 250 Hz square wave on Track 4. The level of the 250 Hz fundamental is ________ volts rms. The SLM is set at 110 dB. In conjunction with the calibration tape, the signal can be used to establish acoustic sound pressure levels." Reconnect cable from hardwired item.

5. Disconnect hardwired item from field end of cable.

6. Reconnect the DGM. Start recording and announce into the DGM: "I am speaking directly into the DGM, serial number _______, at Site _________________." Annotate on commentary track: "A signal is now being recorded on Track 2 in the direct mode. This signal is coming from the DGM, serial number _______. The serial number of the associated SLM is _________________."

7. Disconnect the DGM.

8. Connect the hardwired item. Start recording and announce into the hardwired item. "I am speaking directly into a hardwired item, serial number __________ at Site ________." Annotate on commentary track: "A signal is now being recorded on Track 4 in the direct mode. This signal is coming from the hardwired item, serial number _________. The serial number of the associated SLM is _______________."

9. Reconnect the DGM. Set SLM at 110 dB. Annotate: "The signals which will be recorded come from a General Radio 1562A Sound Level Calibrator. In each

case the level is 114 dB re 0.0002 dynes/cm^2. The signals serve to calibrate the DGM recorded on Track 2. The SLM is set at 110 dB." Record at least 30 seconds of each frequency on the calibrator annotating each as follows: "The present signal is __________ Hz."

10. Annotate: "The personnel at this site are: ______________________

____

____."

11. Describe site, terrain, foliage, local anomalies, etc.; describe weather conditions.

12. Repeat steps 1-4 for each new reel of tape. If aircraft activity precludes immediate calibration and identification it can be postponed but should be done as soon as possible so that it will be near the beginning of the reel.

13. At end of exercise, repeat steps 2-10.

APPENDIX C FIELD EQPT. CHECK LIST

ITEM	Quantity Required	Quantity On Hand	REMARKS
Male BNC to Male BNC 4 ft.	6		
Cables, 1000 ft. cable, 8-pin-Pygmy-to-8-pin-Pygmy	--		should be at site
Cables, 8-pin-Pygmy-to-3-pin-Male XLR	2		
Telephone Wire Reels	--		
Anemometer	--		
CB Walkie-Talkies	1 pair		
CB Battery Packs	--		
CB Short Antenna	--		
IRIG Car Antenna	1		
CB Battery Chargers	--		
Simpson Meter and Leads	1		
Car Battery Jumper Cables	--		needed for Bull Horn
Phone-Plug-to-Dual-Male BNC	3		
Bathroom Tissue	1		
Station Wagon	1		
Car	--		
Tent	--		
Chairs and Cots	2		
Tent Heaters	--		
Colman Stoves	--		
Water Cooler and Cups	1		
Ice Chest	1		
Safari Light and Batteries	--		
Flash Lights	1		
Nylon Rope	--		should be at site
Small Block and Tackle	--		
Machete	--		
Ax	--		
Truck Tow Line	--		

ITEM	Quantity Required	Quantity On Hand	REMARKS
Shovel	--		
Lead Sinkers	--		
Table	--		
First Aid Kit	1		
Filament Tape	1		
Black Electrical Tape	1		
Masking Tape	1		
Label Tape	20		
Black Ballpoint Pens	5		
Tape Reel Labels	20		
Rubber Stamp & Ink Pad (tape box)	--		
Pocket Compass	1		
50 ft. Tape	1		
Tool Box w/ Hand Tools	1		
Polaroid Camera & Film	--		
Clip Boards	1		
Spare Connectors	--		
Recorder Manual	--		
SLM-GR 1551 Manual	--		
SLM-GR 1565A Manual	--		
GR P40 Manual	--		
BATTERIES:			
1-1/2V "D" (SLM-GR 1551-C)	2		
1-1/2V "C" (SLM-GR 1565A)	1		
9V (Calibrator-GR 1562A)	1		
30V (Calibrator-CBS Square Wave)	1		
22-1/2V (Pre-Amp GR P40)	2		
7-1/2V (Monitor)	--		
67-1/2V (SLM-GR 1551-C)	1		

ITEM	Quantity Required	Quantity On Hand	REMARKS
Recorder, 1/2"	--		
Recorder, 1/4"	1		
Nicad Battery Packs for Tape Recorder	2		
Charger for Nicad Battery Packs	--		
Recorder Power Supply and Cables	--		
Empty Reels, 1/2"	--		
Empty Reels, 1/4"	1		
Recording Tape, 1/2"	--		
Recording Tape, 1/4"	10		
Cables, Recorder-to-Car-Batteries	--		
Q-tips and Alcohol	--		
Head Demagnetizer	--		
Spare Belts for Recorders	1		
Sound Level Meters GR1551	2		
Sound Level Meters GR1565A	1		
Microphones GR P5	1		
Preamplifiers and Housings	1		
Sound Level Calibrator 1562A	1		
Square Wave Calibrator	1		
Data Sheets	40		
Commentary Microphone Switch Box	1		
Cable-male BNC to edge track input with built-in attenuator	--		
Monitor / IRIG Radio & Battery Pack	1		
Headphones - Koss low Z	1		
Monitor Box - Amplifier w/ Selector Switch	1		
Cables, Male BNC to Male BNC 2 ft. long	--		
BNC "T"	8		
Hardwired Item	1		
Battery-Powered Oscilloscope	--		
Battery Power Cables for Scope	--		
A.C. Power Cables for Scope	--		

SECRET NOFORN
(WHEN FILLED IN)

APPENDIX D FIELD LOG

Field Operators Data Sheet Page ______ of ______

TEST ____________ LOCATION ____________ DATE ____________

OPERATORS ____________ ____________ ____________
(TEAM LEADER)

REEL NUMBER ______

		Serial # if any	
Tape Recorder (Type)			
Sound Level Calibrator	General Radio 1562A		
Square Wave Calibrator	CBS		
Microphone #1	Microphone General Radio P5		
	Preamplifier General Radio P40		
	SLM General Radio 1551C		
	Length of Cable to Sound Level Meter		feet
	Height of Microphone above ground		feet
	Rainshield or Windscreen, if any		(type)
Microphone #2	Microphone General Radio P5		
	Preamplifier General Radio P40		
	SLM General Radio 1551C		
	Length of Cable to Sound Level Meter		feet
	Height of Microphone above ground		feet
	Rainshield or Windscreen, if any		(type)
Microphone #3	Microphone General Radio P5		
	Preamplifier General Radio P40		
	SLM General Radio 1551C		
	Length of Cable to Sound Level Meter		feet
	Height of Microphone above ground		feet
	Rainshield or Windscreen, if any		(type)
Peak Levels	Impact Analyzer General Radio #1556B		
	Sound Level Meter General Radio #1565A		
Recording Speed			ips

TRACK	FM	DIR	INFORMATION
1			
2			
3			
4			
5			
6			
7			
8			

CAL. FREQ'S	SOUND LEVEL METER READINGS					
	MIC #1		MIC #2		MIC #3	
	ACTUAL LEVEL	SLM SETTING	ACTUAL LEVEL	SLM SETTING	ACTUAL LEVEL	SLM SETTING
400 Hz	dB		dB		dB	
2000 Hz	dB		dB		dB	
1000 Hz	dB		dB		dB	
500 Hz	dB		dB		dB	
250 Hz	dB		dB		dB	
125 Hz	dB		dB		dB	

Terrain ____________

Weather ____________

(over)

SECRET NOFORN
(WHEN FILLED IN)

TRIAL OR DEVICE	COMEX	FINEX	SLM SETTINGS			PEAK METER READG		
			#1	#2	#3			
						db		

SECRET NOFORN
(WHEN FILLED IN)

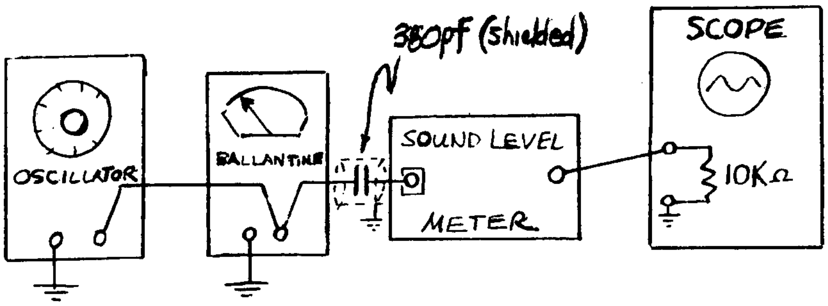
BEFORE USE EQUIPMENT CHECKOUT & CALIBRATION FOR G.R. SOUND LEVEL METER MODEL 1551-CS2 SERIAL No.________________	CLD- 5025-TR-002 Page 1 of 5 Pages **APPENDIX E(1)**	

Test No.	Description	Results	Initial Date
1	This procedure covers the following items of checkout on the Sound Level Meter. 1. Batteries 2. Gain, Attenuator, Frequency Response 3. Distortion 4. Noise 5. Microphonics		
1.1	Batteries		
1.1.1	Turn SLM on by raising microphone stand.		
1.1.2	Turn Switch to Fil-1 and check to see that the meter reads in Bat area.	Yes ___ No ___	
1.1.3	Turn switch to Fil-2 and check to see that the meter reads in Bat area.	Yes ___ No ___	
1.1.4	Turn Switch to PL and check to see that the meter reads in Bat area.	Yes ___ No ___	
1.1.5	Turn attenuator to 130/cal position. Turn weighting switch to CAL. Turn CAL thumbset control to see if a <u>full scale</u> reading can be obtained.	Yes ___ No ___	
1.1.6	With a Simpson Meter measure the voltage between pin 2 and pin 1 on the microphone stand. It should read 40 volts ± 5 volts Notes: Battery life expectancy: <u>A Batteries</u> (Filament), 6 to 7 days <u>at 8 hours/day</u>. <u>Plate Batteries</u>, 12 to 14 days <u>at 8 hours/day</u>.	______ Volts	
1.2	<u>Amplifier Gain, Attenuator, Frequency Response</u>		
1.2.1	Connect the equipment as shown in Figure 1. Note that it is very important to have the capacitor and load resistor as shown.		

BEFORE USE EQUIPMENT CHECKOUT & CALIBRATION FOR
G.R. SOUND LEVEL METER MODEL 1551-CS2

CLD- 5025-TR-002
Page 2 **of** 5 **Pages**

SERIAL # ______________________

Test #	Description	Results	Initial Date
	 FIGURE 1		
1.2.2	Set the Sound Level Meter to: Attenuator = 130 Weighting = 20 KHz Meter = Fast		
1.2.3	Turn the oscillator to 1 KHz and increase its level such that the Ballantine meter reads 1.0 volts.		
1.2.4	Turn CAL control on the Sound Level Meter to read +6 on 130 attenuator on the Sound Level Meter.		
1.2.5	Adjust the oscilloscope to obtain a good waveform. Then measure the output voltage on the scope. It should read 1.6 $\pm$ 3 volts peak-to-peak. The waveform should be a sine wave.		
1.2.6	Turn the frequency knob on the oscillator to 30 Hz. Then sweep the oscillator very slowly from 25 Hz to 12KHz. There should be less than 3 dB change on the Sound Level Meter.	Max______dB Min______dB Change______dB	
1.2.7	Turn the oscillator to 1KHz and reduce the voltage by 10 dB. Switch the Sound Level Meter attenuator to the next lower scale. The SLM meter should read +6±5 dB. Repeat step 1.1.6 and continue this process until the range switch is at 40. Note that the waveform on the scope is a good sine wave at each setting.		

BEFORE USE EQUIPMENT CHECKOUT & CALIBRATION FOR
G.R. SOUND LEVEL METER MODEL 1551-CS2

CLD- 5025-TR-002
Page 3 of 5 Pages

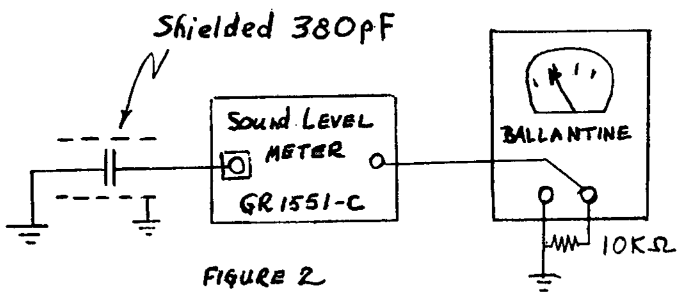
SERIAL # ______________________

Test #	Description	Results	Initial Date
	Atten Setting	(1KHz) Meter Rdg. / dB Change	
	120	____ dB ____ dB	
	110	____ ____	
	100	____ ____	
	90	____ ____	
	80	____ ____	
	70	____ ____	
	60	____ ____	
	50	____ ____	
	40	____ ____	
1.3	Distortion Measurement		
1.3.1	Connect the Distortion Analyzer to the output of the SLM. Return the attenuator of the Sound Level Meter to 130. Turn the oscillator to 100 Hz and bring up the level to read +10 dB (full scale) on the SLM meter.		
1.3.2	Turn the Distortion analyzer to Set Level and the attenuator knob to 100%. Adjust the Input Control to give full scale on the Distortion Analyzer Meter.		
1.3.3	Turn to Distortion and adjust the frequency dial to 100 Hz and null the meter. Switch the attenuator to 30% and null using the frequency dial. Repeat on 10% and 3% settings. Once the null is obtained adjust the balance knob for a minimum reading. Read the percent distortion. It should be less than 1.25%.	SLM Set at 130 100 Hz ______%	
1.3.4	Turn the oscillator to 1000 Hz and repeat steps 1.3.2 and 1.2.3.	1KHz, ______%	
1.3.5	Turn the oscillator to 10 KHz and repeat steps 1.3.2 and 1.3.3.	10 KHz, ______%	

BEFORE USE EQUIPMENT CHECKOUT & CALIBRATION FOR
G.R. SOUND LEVEL METER MODEL 1551-CS2

CLD- 5025-TR-002
Page 4 **of** 5 **Pages**

SERIAL #

Test #	Description	Results	Initial Date
1.3.6	Reduce the input voltage and switch the SLM attenuator to 60. Then bring the oscillator up in level such that the SLM meter reads +10 dB.		
1.3.7	Repeat steps 1.3.2 through 1.3.6.	SLM Set at 60 100 Hz ____% 1 KHz ____% 10 KHz ____%	
1.4	Noise Measurement		
1.4.1	Connect the equipment as shown in Figure 2.  FIGURE 2		
1.4.2	Switch the SLM to 40. Place the Ballantine on the 0.1 volt scale. The reading should be at least 4 dB below full scale.	-________dB (below full scale)	
1.4.3	Switch the SLM to 50. The Ballantine should read more than 14 dB below full scale.	-________dB (below full scale)	
1.4.4	Switch the SLM to 60. Switch the Ballantine to 0.01 volt scale. The Ballantine should read below full scale.	-________dB (below full scale)	

BEFORE USE EQUIPMENT CHECKOUT & CALIBRATION FOR
G. R. SOUND LEVEL METER MODEL 1551-CS2

SERIAL # ______________________

Test #	Description	Results	Initial Date
1.4.5	Repeat Step 1.4.3 except increase the Sound Level Meter in 10 dB steps. The meter should read more than 10 dB below full scale on the 0.01 volt setting for each position above 60 on the SLM.	70, - dB (below Full Scale) 80, - dB 90, - dB 100, - dB 110, - dB 120, - dB 130, - dB	
1.5	Microphonics		
1.5.1	Place the SLM in the 40 position, gently tap its case on all sides. The meter should not bounce above 0 dB.	______ dB (max meter swing)	

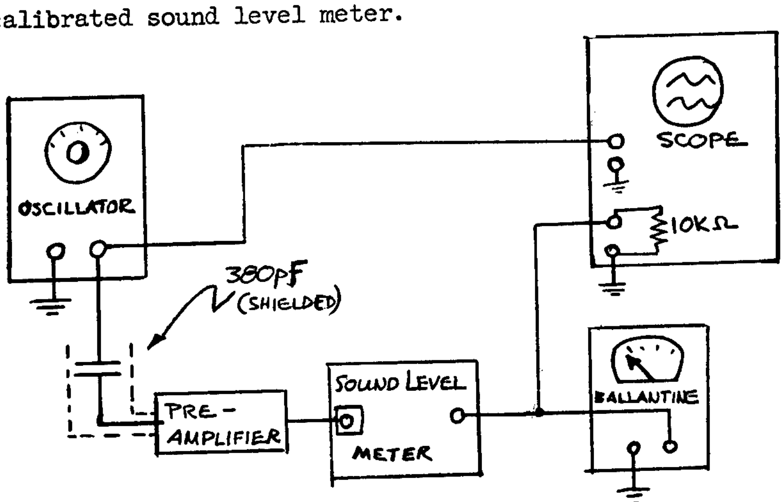
EQUIPMENT CHECKOUT AND CALIBRATION FOR
G.R. PREAMPLIFIER MODEL 1560-P40

CLD- 5025-TR-003
Page 1 of 3 Pages

SERIAL # ______________

APPENDIX E(2)

CONTRACT # ______________ PROJECT # ______________

Test #	Description	Results	Initial Date
1.	This procedure covers the following items of checkout on the preamplifier: 1. Gain 2. Response 3. Distortion 4. Noise		
1.1	Gain, Response		
1.1.1	Connect the equipment as shown in Figure 1. Use only a calibrated sound level meter.  FIG 1		
1.1.2	Switch the preamplifier to the X1 position. Turn the sound level meter to the 120 dB setting. Place the voltmeter in the 1.0 volt range.		
1.1.3	Set the oscillator to 1 kHz and adjust the level control so that a 1.0 volt peak-to-peak signal is read on the upper beam of the scope. Observe that the lower beam gives a wave that is similar in shape to the upper beam. It should be a 1.5 ± 0.3 volt peak-to-peak level.	Good Waveform ___Yes ___No ____ V p to p	

EQUIPMENT CHECKOUT AND CALIBRATION FOR
G.R. PREAMPLIFIER MODEL 1560-P40

CLD- 5025-TR-003
Page 2 of 3 Pages

SERIAL # ____________________

CONTRACT # ____________________ PROJECT # ____________

Test #	Description	Results	Initial Date
1.1.4	Record the dB reading on the voltmeter. Then very slowly sweep the oscillator from 30 Hz to 12 kHz. There should be no more than a 3 dB variation over this range.	+____ max. dB reading ____ min. dB ±____ dB change	
1.1.5	Switch the SLM to 130 attenuator setting. Increase the oscillator voltage so that the SLM reads full scale. There should be no noticeable clipping on the lower scope waveform. (Note: This is the maximum useful level of the preamp. Any increase in input voltage will cause waveform clipping.)	Clipping at 140 dB (+10 on the 130 scale) ___Yes ___No	
1.2	Distortion		
1.2.1	With the same setup as Figure 1, connect the Distortion Analyzer to the output of the SLM. Place the attenuator on the SLM to 120 dB.		
1.2.2	Adjust the frequency of the oscillator to 100 Hz and bring up the level to give full scale (+10 dB) on the SLM. Switch the Distortion Analyzer to set level and 100%.		
1.2.3	Adjust the level control on the Distortion Analyzer to read full scale on its meter. Switch to Distortion and turn the frequency knob to 100 Hz. This will null out the signal. Turn to 30%, 10% and then 3% while nulling the meter using the frequency knob. Then adjust the balance control for the lowest meter reading (nulled).		
1.2.4	The distortion measurement should be lower than 1.25%.	100 Hz ____% Distortion	
1.2.5	Turn the oscillator frequency to 1 kHz. Then repeat Step 1.2.3. The distortion measured should be less than 1.25%.	1 kHz ____% Distortion	
1.2.6	Turn the oscillator frequency to 10 kHz. Repeat Step 1.2.3. The distortion should be less than 1.25%.	10 kHz ____% Distortion	

EQUIPMENT CHECKOUT AND CALIBRATION FOR
G.R. PREAMPLIFIER MODEL 1560-P40

CLD- 5025-TR-003
Page 3 of 3 Pages

SERIAL NO. ____________

CONTRACT NO. ____________ PROJECT NO. ____________

Test No.	Description	Results	Initial Date
1.2.7	Reduce the oscillator voltage to minimum and switch the SIM to 60. Then increase the oscillator voltage so that the SIM reads full scale (+10 dB).		
1.2.8	Repeat Steps 1.2.3 through 1.2.7. The percent distortion should be less than 1.5%.	100 Hz ____ % 1 kHz ____ % 10 kHz ____ %	
1.3	Noise		
1.3.1	Short circuit the input capacitor to ground.		
1.3.2	Switch the SIM attenuator to 60. Place the Ballantine meter on the 0.01 volt scale. The meter should read below full scale.	____ dB below full scale	

BEFORE USE CHECKOUT PROCEDURE FOR GR 1560-P5 MICROPHONE

CLD- 5025-TR-005
Page 1 of 3 Pages

SERIAL # ____________________

APPENDIX E(3)

Test #	Description	Results	Initial Date
1.	This procedure covers the following items of checkout: 1. General sensitivity and Response. 2. Reciprocity calibration.		
1.1	General calibration using GR 1562A calibrator.		
1.1.1	Plug the microphone into a calibrated sound level meter, or into a calibrated pre-amplifier and Sound Level Meter combination. Turn on equipment by raising the mic holder, and switch the attenuator to 110 scale.		
1.1.2	Start the 1562A calibrator by first turning the knob counterclockwise for 1 second and observe the battery indicator light. Then switch to 1 KHz. Listen to see if a tone is heard. Make sure the large hole adapter is plugged into the calibrator.		
1.1.3	Slowly place the calibrator on the microphone, (rapid movement will cause excessive pressure on the microphone diaphragm and destroy the unit). The SLM should read near +4 dB on the 110 setting. If it is close turn the CAL knob for a +4 dB reading.		
1.1.4	Turn the Calibrator to 2KHz and record the dB reading. Repeat for the other frequencies. There should be less than ± 1 dB variation over the frequency range.	2KHz ______ dB 1KHz +4 500 Hz ______ 250 Hz ______ 125 Hz ______	
1.2	Reciprocity Calibration		
1.2.1	Connect the equipment as shown in Figure 1.		

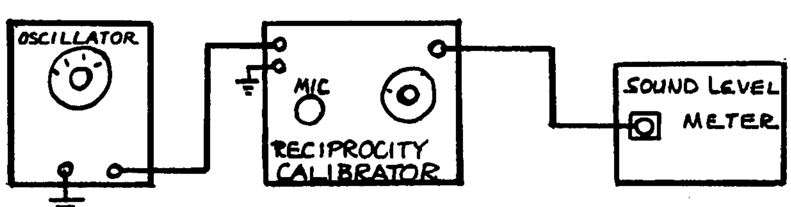


FIG 1

BEFORE USE CHECKOUT PROCEDURE FOR GR 1560-P5 MICROPHONE

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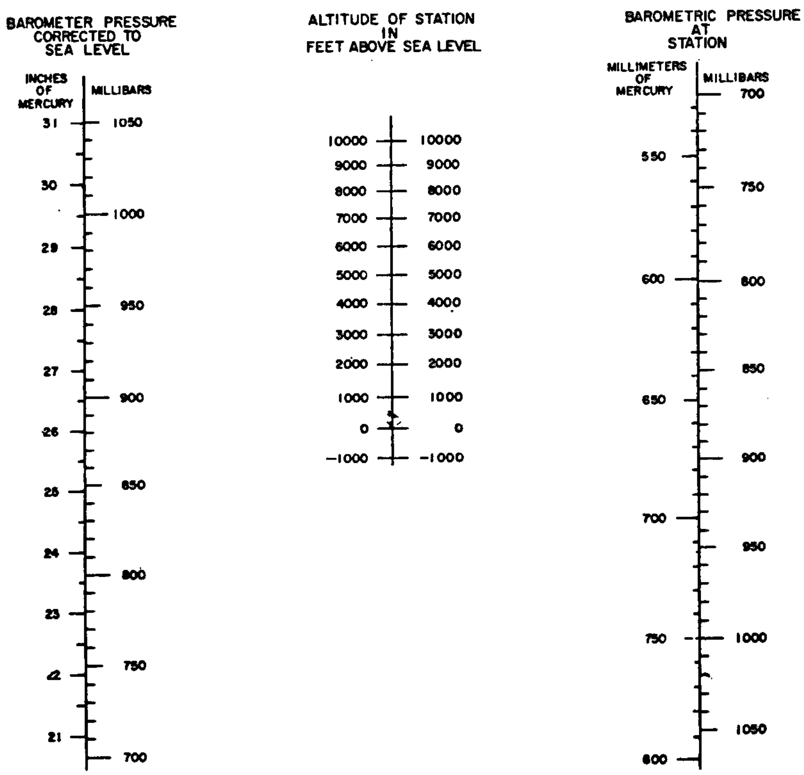
Test #	Description	Results	Initial Date
1.2.2	Place the microphone very slowly into the Reciprocity Calibrator and connect the cable. (See warning 1.1.3)		
1.2.3	Place the controls in the following positions: Start 92 dB Sensitivity = Normal Mic Current = Normal		
1.2.4	Call the local airport, or weather station, and obtain the correct barometric pressure. Then correct the reading for altitude changes according to the following table. Set the red dot on the large dial on the reciprocity calibrator next to this value on the plastic scale.		



Nomograph for applying altitude correction to barometric pressure. Place straight-edge across proper points on center and left-hand scales and read actual pressure on right-hand scale.

BEFORE USE CHECKOUT PROCEDURE FOR GR 1560-P5 MICROPHONE

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Test #	Description	Results	Initial Date
1.2.5	Place the oscillator on 1KHz. Turn the knob to Read 1 and adjust the oscillator for an on scale reading on the Reciprocity Calibrator meter. Also turn the attenuator on the Sound Level Meter to obtain a reading. Note this reading.		
1.2.6	Then turn to Adj 1 and rotate the large knob to obtain the same Sound Level Meter reading as in step 1.2.5.		
1.2.7	Turn to Read 2 and note the SLM meter reading. Turn to Adj 2 and adjust the large knob to match the same SLM reading on Read 2.		
1.2.8	Repeat 1.2.7 for Read 3 and Adj 3.		
1.2.9	Repeat 1.2.7 for Read 4 and Adj 4.		
1.2.10	The microphone sensitivity is now obtained by reading the number under the hairline onthe outer large dial. Repeat steps 1.2.4 through 1.2.10 for the frequencies listed below. 25 Hz ____________ dB re 1 volt/microbar 1 KHz ____________ dB re 1 volt/microbar 5 KHz ____________ dB re 1 volt/microbar If the sensitivities of the microphones differ from -62 dB by more than ±3 dB at any frequency, the microphone is faulty		

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REPORT DOCUMENTATION PAGE

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1. REPORT NUMBER: NADC-AWG-SU	2. GOVT ACCESSION NO.
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Frequency domain format	Mathematical modeling	Normal mode theory

20. ABSTRACT *(Continue on reverse side if necessary and identify by block number)*

With the ending of American military involvement in the Indochina conflict it became appropriate to summarize the unclassified aspects of those investigations which might have a reference value to future scientific workers both in the civilian and military endeavors. The investigations of the Acoustical Working Group (AWG), a committee of knowledgeable persons from various civilian and Department of Defense agencies responsible for performing acoustical studies relevant to the Indochina conflict, clearly (Cont.)

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fall in this category. They embrace, in general, the study of sound propagation, a subject important to noise abatement, hearing protection, and other ecological and environmental concerns. Some of these studies (more than 50 formal reports were prepared in a period of five years) also are significant in a general academic sense and as such have been selected to appear in this volume. The topics covered are data acquisition, reduction, analysis, acoustical propagation, and the characteristics of sound sources.